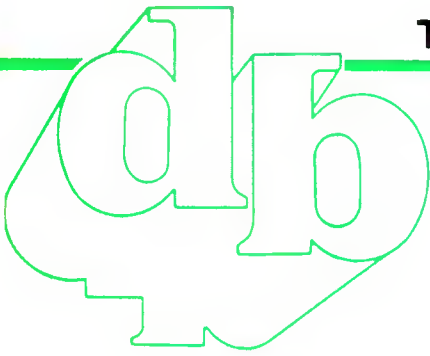


MARCH/APRIL 1988
\$2.95

THE SOUND ENGINEERING MAGAZINE

serving: recording, broadcast and sound contracting fields



Guides: Power Amplifiers
and THE ELECTRONIC COTTAGE





There can be no compromise!

Tour the premier recording studios of the world — from London to New York to L.A. — and you'll find they have one thing in common: "no compromise" recorders from Studer of Switzerland.

Sure, their Studer multitrack mastering decks are a big investment, but you can make an equally sound choice for your production needs for a whole lot less. You can own a two-track production recorder with the same Studer heritage — a machine that has many of the same production features, the same uncompromising audio performance and the same level of manufacturing perfection that has made Studer Revox recorders the world standard — THE REVOX PR99 MKII is the machine!

Like its "big brothers" in the top studios, the PR99 MKII is a professional machine built for long-term perfor-

mance. From the solid diecast aluminum transport chassis and head block to the servo capstan motor and the modular electronics, everything is milled, drilled and mounted with Swiss precision. The parts fit together right — and stay there.

The PR99's professional features are perfect for efficient, accurate tape production: • RealTime counter that reads both plus and minus hours, minutes and seconds; • True Auto Locator allows precise, automatic search-and-cue to any preselected address point; • Zero Locate to return the tape to the zero counter location — EXACTLY! • Auto Repeat to continuously replay a tape segment of any length.

Plus: • Built-in, front-panel varispeed; • Self-Sync; • Input and output mode switching; • Edit mode switch; • Tape dump; • Calibrated and Uncali-

brated "+4" balanced and floating inputs and outputs; • 10½" reel capacity.

As for sound quality, the Studer heritage again allows no compromise. We think you'll find the Revox PR99 MKII to be sonically superior to anything in its price range. Audition the Revox PR99 MKII at your Studer Revox Professional Products Dealer, or contact: Studer Revox America, Inc., 1425 Elm Hill Pike, Nashville, TN 37210; (615)254-5651.



PR99 MKII Real Time Counter and Autolocator.

STUDER REVOX

Circle 15 on Reader Service Card



The broadcast engineer

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• Dave MacCarn, Director of Engineering for WGBH-TV and Radio (at left) and John Voci, Operations Director for WGBH Radio, and the author of this issue's article on WGBH's work in digital transmission to consumers; both are standing in front of equipment used in this operation.

We also want to congratulate WGBH because as a result of the work that is explained in our article, as well as other digital broadcast technology pioneered by them, they have been presented with the Armstrong Award for technical achievement for 1988.

Editorial

Some of you are seeing **db Magazine** for the first time, having picked up the copy you hold at the NAB. As you see, we are a professional audio engineering magazine. Hopefully, you like what you read, and we'll be hearing more from you. I particularly want to you to see and comment on our new *Broadcast Audio* columnist Randy Hoffner. Randy is Senior Staff Engineer at NBC-TV in New York. In addition, our broadcast audio coverage includes our cover story on WGBH in Boston, and Brian Battles' article, *Gain the Ad-Vantage*. John Barilla's *L & R—The Big-Little Production Company* also covers broadcast audio.

Now I want to address myself to our existing base of reader/subscribers, some of whom have been with us since our starting days in 1967. Others have come to us later, but no matter: You know your **db Magazine**. So what is this you have in your hand?

Over the years, as the audio professional has changed, so has **db Magazine's** editorial changed. Be it broadcast audio, recording studios large and small, and performance/sound contracting engineering, what was done twenty years ago, is related to, but not the same, as what we do today. **db Magazine** has been in the forefront of reporting pro audio, and will continue in that fashion.

But it was time to also modernize the look and feel (as it were) of **db Magazine**.

The results are in your hand. Editorial features are up front, columnists are to the back. Our Buyer's Guides are in the middle. That permits us to use editorial space more efficiently, and at the same time, make those pages more attractively usable and functional for you.

There are more changes. As pro audio changes, we change. Beginning with this issue, the feature *The Electronic Cottage* becomes regular. This title will encompass the growing segments of personal studios. They may be small, inexpensive, and in their owner's bedroom; or they may be small, expensive, and in three rooms of the house. But they will all represent a growing phenomenon in audio—the personal studio, operated by its owner for specialized use. *The Electronic Cottage*—what the pro audio industry will be.

This year, 1988, will see many other exciting new technologies. For broadcasters and recording studios, the emerging R-DAT professional systems that we've already written about are just the beginning. In just a few years these first 2-channel R-DATs will become 8-channel R-DATs. Imagine having an 8-channel digital recorder whose battery supply may well weigh more than the machine!

Meanwhile, analog technology takes a quantum leap for the smaller studio market with Tascam's introduction of an 8-track analog cassette system. Read all about that unit in our NAMM Roundup in this issue.

Is **db Magazine** for you? If you are in professional audio, and want to learn all about the latest technologies, all you need do is read our pages.

L.Z

Editor/Publisher
Larry Zide

Associate Publisher
Elaine Zide

Editorial Assistant
Carol A. Lamb

Technical Editor
Corey Davidson

Contributing Editors
Bruce Bartlett
Mark E. Battersby
Brian Battles
Drew Daniels
Len Feldman
Randy Hoffner

Graphics & Layout
Karen Cohn

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Trademarked names are editorially used throughout this issue. Rather than place a trademark symbol next to each occurrence, we state that these names are used only in an editorial fashion and to the benefit of the trademark owner, and that there is no intention of trademark infringement.

The first 500-Hz Driver that doesn't turn cymbals into trash can lids



Listen to most of today's HF drivers, including our leading competitor's, and you could logically conclude that "trashy" sound is an inescapable fact of life. Poor definition, inadequate output beyond 10 kHz, annoying breakups, and "ringing" are all too common.

EV engineers, rejecting the notion that poor high frequency sound is inevitable, created the DH1A, a driver that deals effectively with every one of these problems.

To boost high-frequency output we utilized a magnet with the greatest flux density available, plus an optimized, balanced magnetic circuit to "stiffen" the coupling between the amplifier and the diaphragm. The resulting increase in high-end response also solved the problem of definition and articulation, so the sound is cleaner and

livelier, with better transient-handling capability. As a result, trashy instrumental and vocal sounds are consigned to the trash can, where they belong.

The 10 kHz breakup you've heard in our competitor's driver was eliminated by using a 3-inch diaphragm instead of the other guy's 4-inch component, moving the primary diaphragm breakup point all the way out to 16 kHz, well beyond fundamentals and first harmonics.

A field-replaceable diaphragm, we reasoned, could make the DH1A still more useful. So we made that a part of the package, too. Plus the option of 8- and 16-ohm impedance

match. And our EV-exclusive PROTEF™ feature that guards against voice coil damage.

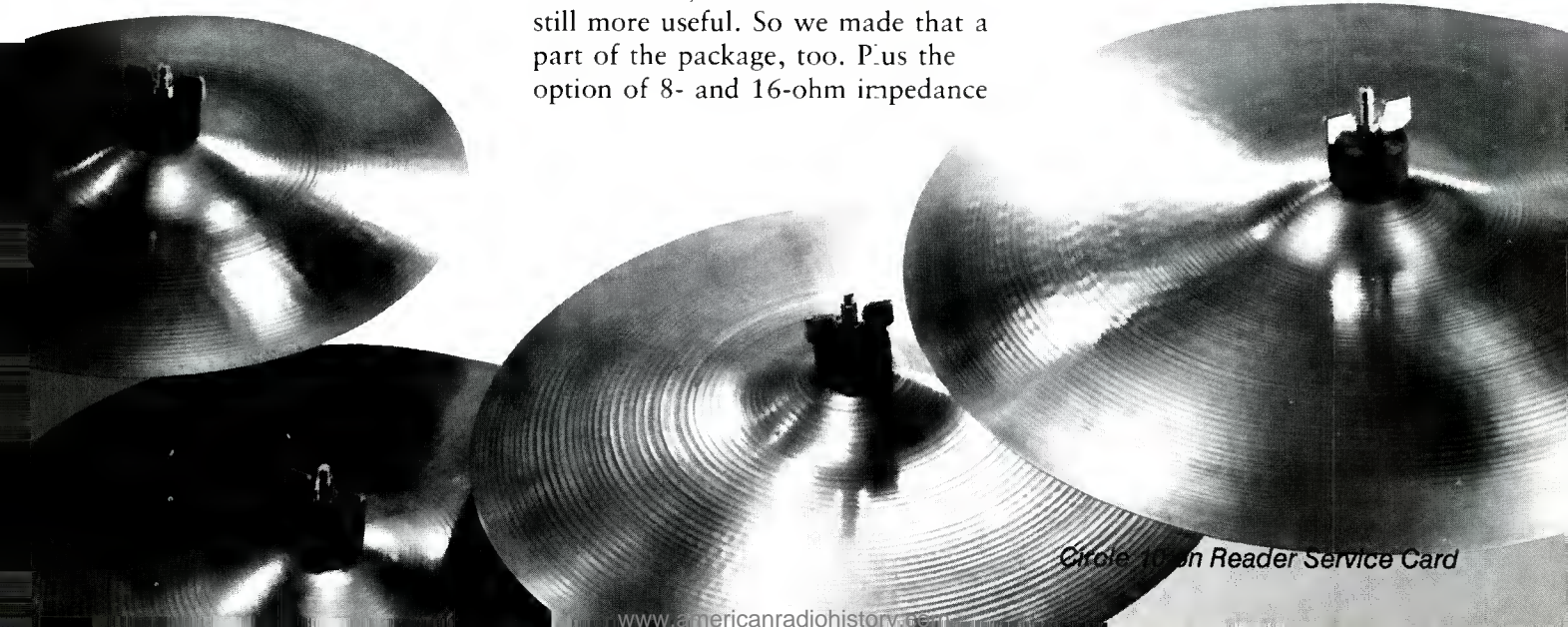
Talk, as they say, is cheap.

So, we insist that you make us prove our claims. Audition a DH1A today and hear for yourself how easily you can bid a hasty goodbye to trash-can cymbals and high-end distortion.

For more information, write Electro-Voice, Inc., 600 Cecil Street, Buchanan, MI 49107.



Electro-Voice®
SOUND IN ACTION™



Circle 10 on Reader Service Card

Its features are that sounds good

Sony is pleased to announce a merger. Analog recorders that combine the advanced features you need with the quality sonics you crave: the remarkable APR-5000 series.

Whether you need a recorder with a genius for post-production or one for high-quality studio mastering, there's an APR-5000 that fits.

Their 16-bit microprocessor controlled transports handle tape smartly, yet gently. And "intelligent" head assemblies make changing head formats a snap.

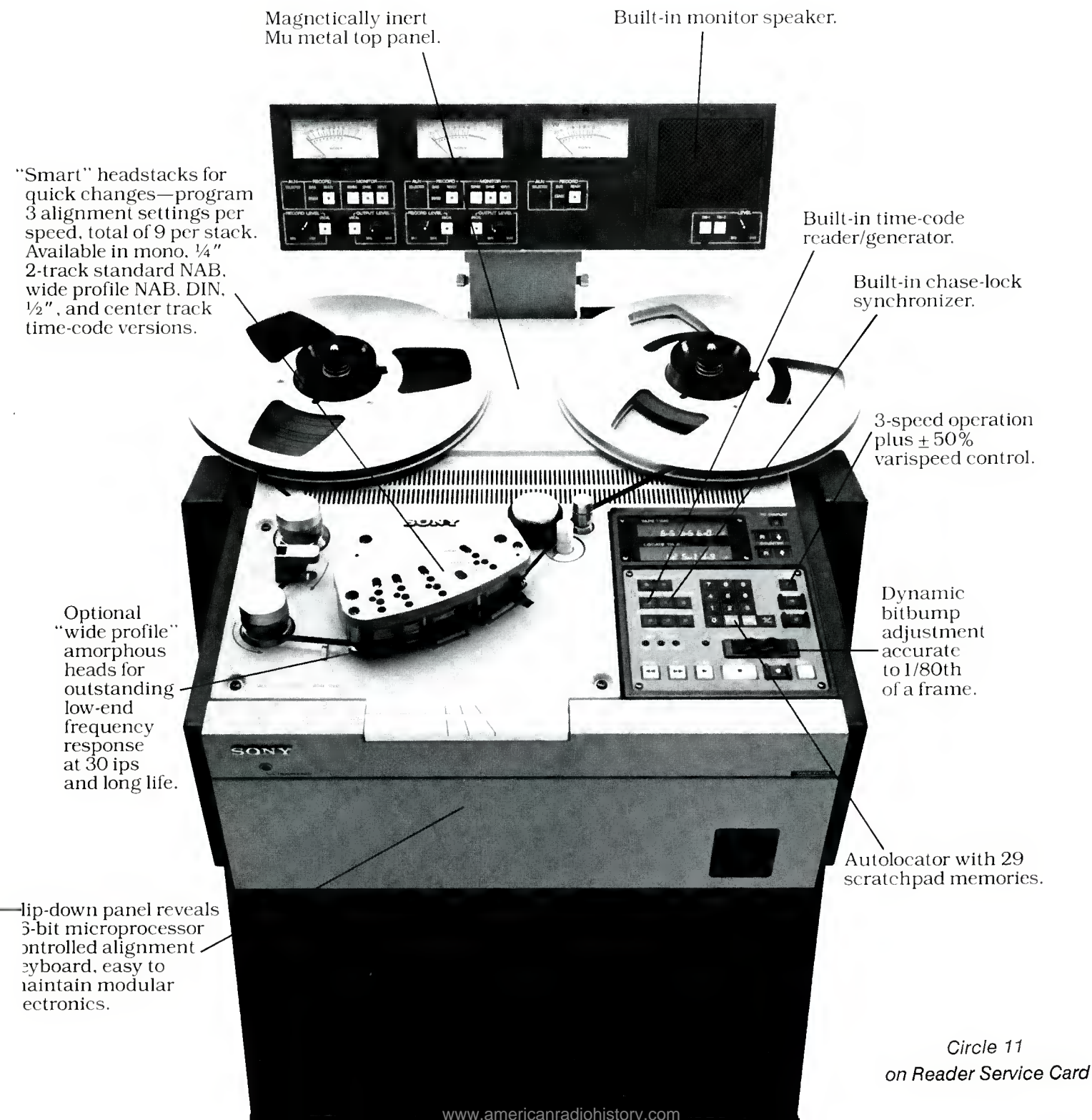
And when it comes to sound quality, transformerless design and 400 kHz bias enhance high-end performance. While optional "wide profile" heads create extended low frequency response at 30 ips.

So, if you've been waiting for a precision analog recorder that finally breaks the sound barrier, don't wait. Contact your Sony Professional Audio representative. Or call Sony at 800-635-SONY.

SONY®

Professional Audio

n't the only thing



Circle 11
on Reader Service Card

Letters

Dear Editor,

I enjoy the entire magazine from cover to cover. More broadcast engineers should take the time to go through db. Our production people almost fight to read Brian Battles' Ad Ventures.

Jim Mross Jr.

Chief Engineer KIXX/KTCL

What can we say, we know a lot of db Magazine readers are broadcasters, but thanks to Jim, it's said clearly. Ed.

THE AMERICAN HEART
ASSOCIATION
MEMORIAL PROGRAM



WE'RE FIGHTING FOR YOUR LIFE



American Heart Association

This space provided as a public service.



REMOVES VOCALS FROM RECORDS!

Our VOCAL ELIMINATOR can remove most or virtually all of a lead vocal from a standard stereo record and leave most of the background untouched! Record with your voice or perform live with the backgrounds. Used in Professional Performance yet connects easily to a home component stereo system. Not an equalizer! We can prove it works over the phone. Write or call for a free brochure and demo record.

Listen..

Before You Buy!

- Time Delay
- Reverberation
- Crossovers
- Noise Reduction
- Compressor/Limiters
- Expanders
- Spectrum Analyzers
- Parametric EQ

Don't have regrets about paying too much for a lesser product. In demos and comparisons, we'll show you why we're Better! Our Factory Direct sales allow us to produce a Superior product and offer it to you at a Lower price. Call or write for a free full length Demo Album and 24 page brochure.

LT Sound, Dept. DB-7, 7980 LT Parkway
Lithonia, GA 30058 (404) 482-4724
24 HOUR PHONE DEMO LINE: (404) 482-2485

Calendar

• A four-week program, comprised of six accredited graduate level courses in acoustics and signal processing, will be offered in June 1988 by Penn State's Graduate Program in Acoustics in cooperation with the University's Applied Research Laboratory. Courses offered include: Fundamentals of Acoustics, Underwater Sound Propagation, Sonar Engineering, Digital Signal Processing, Electroacoustic Transducers, and Acoustical Data Measurement and Analysis.

For more information contact:

Dr. Alan D. Stuart

Summer Program Coordinator

Penn State Graduate Program in Acoustics

PO Box 30

State College, PA 16804

• The upcoming schedule for the SYNERGETIC AUDIO CONCEPTS two-day audio engineering seminars is as follows:

New York City area- April 22-23

Nashville- May 3-4

Toronto area- June 23-24

Syracuse- June 28-29

"Master Loudspeaker Designer's Workshop," conducted by Dr. Eugene Patronis and staff, will be held in Atlanta, GA on April 15-17 (please note change of date.)

For more information contact:

Synergetic Audio Concepts

PO Box 1239

Bedford, IN 47421

• The United States Institute for Theatre Technology, Inc. (USITT) is presenting the annual conference and Stage Expo '88 commercial exhibit show. The event is being held at the Disneyland Hotel in Anaheim, California from March 23-26, 1988. Stage Expo attracts suppliers and manufacturers of those products needed for live performances.

For further information contact:

RJA Exposition Management

486 Fullerton Ct

San Jose, CA 95111

(408) 225-6736

• The third edition of the Magis Exhibition of Equipment and Technology for Theatres and Cinemas will be held from March 22-25, 1988 at the Rimini Trade Fair Centre in Italy. The exhibition will include sound and P.A. systems, special effects, stages and mobile structures, and electronic editing and dubbing equipment. Promotion is aimed at technicians involved in the installation and hire of equipment for cinemas, theatres, shows and concerts, but also at lighting designers, cinema photography directors, set designers, show organizers and owners and managers of theatres and cinemas. The sixth edition of SIB (International Exhibition of Technology for Dance Venues) will be held from March 22-25, 1988, also at the Rimini Trade Fair Centre in Italy.

• The Electronic Music Workshop will be held June 20-24, 1988, at New England Conservatory in Boston, Massachusetts. It will consist of lecture demonstrations followed by hands-on experience with a wide variety of both analog and digital synthesizers. No prior knowledge of electronic music is assumed or required. The workshop is available for undergraduate/graduate credit or non-credit. Robert Ceely, who is offering this workshop, is director of the Electronic Music Studio at New England Conservatory. Contact the New England Conservatory (290 Huntington Ave, Boston, MA 02115) for further information.

• The Professional Education and Training Committee of the International Communications Associations (ICIA) is offering regional training seminars for the audio-visual and computer industries. The seminars will provide current information on the newest applications and techniques available on computer interfacing for projection and monitor displays. The dates are:

Los Angeles: March 31-April 2

Chicago: April 7-9

New York: August 11-13

Toronto: September 1-3

Atlanta: October 6-8

For more information, contact ISIA (3150 Spring St, Fairfax, VA 22031-2399).



Microphones

In the studio, over the air or up on stage, there's a Fostex RP mic specifically designed for the job at hand. RP stands for regulated phase, a transducer technology which has been awarded over 20 international patents to date. These mics have the warmth of condensers, the ruggedness of dynamics and a sound as transparent as it gets.



Headphones

These are more outstanding examples of RP Technology. Model T-20 has become almost legendary among studio musicians, producers and engineers. Its flat response at any listening level and its comfortable design help you listen longer without fatigue. And the sound is so clear and well-defined, critical listening is enjoyable.



Speaker Systems

You're up & going with Fostex PA systems. Modular designs let you control the sound according to the needs of the event. Stack them, gang them. From a simple portable PA to an entire rig, look to Fostex speaker systems to help you solve your sound problems.



Powered Mixers

Model MP-800 has 8 inputs and delivers 180 W per channel and Model MP-1200 has 12 inputs and delivers 250 W per channel. These rugged, road worthy stereo mixers have quiet running fans, digital echo, normalled connections at all critical patch points, stereo graphics on the mains and super monitoring flexibility. The best at any price.



Complete PA Systems

Look to Fostex for any and all of your PA needs. Complete systems or individual components. High quality sound from input to output.

Fostex
Pro Sound Division

15431 Blackburn Ave., Norwalk, CA 90650
(213) 921-1112

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Circle 12 on Reader Service Card

Gain the Ad-Vantage

HERE IN THESE PAGES MY JOB IS NORMALLY TO SUGGEST to owners of small recording studios ways to write, produce and prepare radio commercials. I also try to help them find ways to locate and land clients, and I give them technical facts about the requirements of radio stations. To tie in with the NAB show, however, this article is intended for those who have to put the ads on the air—radio station personnel, production directors, disk jockeys, engineers and any other people whose jobs require them to take spots that the sales department brings in and get them into the air studio, ready for playing.

I spent over a decade of my broadcast career as a production director at various radio stations, and as a D.J. most of that time, too, so I believe I've seen, heard and created just about every possible configuration of advertisement. There are spots ranging from excellent to embarrassing—tapes that come in on digital stereo reel-to-reel or on K-Mart brand cassettes. Sponsors submit ads that were produced at the world's foremost recording studios as well as spots they've taped in the living room on their Yorx cassette recorder using department store mics. (By the way, you can recognize these "amateur hour" ads because they usually have the client's lisping adolescent daughter as announcer in one channel with a Kenny Loggins or Bon Jovi tune blaring on the other channel.)

I try to explain the technical basics dictated by radio station facilities so that the readers of *db* will not be responsible for adding to the pile of sludge and sorry garbage received by stations each day. Why do I do it? Two reasons: 1) I have been on the receiving end of a lot of rubbish, and 2) I want to help the studio owner/entrepreneur get off on the right foot. Unfortunately, everyone who tries their hand at creating radio commercials doesn't see my column (no kidding!), and there's always going to be miles of dreadful tape rolling into your "IN" basket.

On the other hand, I also know how many radio stations are pitifully lame in terms of production equipment. There is also the sad fact that broadcasting is loaded with ignorant, untrained or apathetic individuals who operate the production gear. There is also a dismal absence of quality standards enforced at stations respecting maintenance and operational practices. We'll look at these issues and recommend ways to optimize the sound of the commercials that get to the control room at a normal radio station. (I think that last phrase is an oxymoron).

STATION EQUIPMENT

Let's start by letting some readers off the hook; I know there are a few dozen radio stations that are fully equipped with modern multi-track (and even digital) tape machines,

huge mixing consoles, tens of thousands of dollars' worth of signal processing and special effects devices, exceptional microphones, state-of-the-art tape cartridge decks, Dolby or dbx noise reduction, and all other manner of high-tech facilities. These stations also have the budgets to afford full-time recording or maintenance engineers whose duties automatically include tape head cleaning and alignment, level adjustment, testing, repair, and general housekeeping and maintenance chores. In some cases, the talent (D.J.s) isn't even allowed to lay hands on the machines. If you work at a station in this rare category, you probably won't get very much out of this article. In fact, you probably have "people" who do your production for you.

However, since most of us don't work at one of these broadcast showcases (for the moment), you'll inevitably have to deal with a situation that is familiar to most radio people. The majority of stations' production studios are outfitted with elderly reel-to-reel machines, tape-gobbling cart decks, World War II vintage mics, and very little else (all occupying a closet with walls covered with egg cartons for "acoustical treatment"), we are going to find out how you can make the most of your facility.

In all the years I have spent in radio, I have learned one primary lesson: Owners don't care about equipment. As long as it puts out a listen-able signal with minimal down time, as long as the gear doesn't get stolen, as long as the ratings are competitive, the proprietors of radio stations don't give an aerial act of amorous anastomosis about audio quality, especially since it costs money. To paraphrase Nathaniel Hawthorne, with regard to studio equipment expenditures, station owners are usually as close as a vice.

You may recycle old reels of public affairs shows or the freebie religious programs for mastering tape; you might need to use a book or an elastic band to keep the mic stand from collapsing to the table; perhaps you sort tape cartridges in bins labeled "Good (use for music and IDs)," "Broken—do not use," and "O.K. just for commercials." I know how it is. Don't despair. I cut some of my favorite works, even award winners, in wretched hellhole production rooms. You can do it.

The most important thing to remember about radio commercial production is that great spots come from *you*, not from your equipment. Certainly there are some limitations on how easily you can get certain sounds, and the speed at which you can work, but most of the all-time classic advertisements I have ever heard were put together in the days before Harmonizers, digital delays, computerized effects and MIDI. This is because copywriters and engineers had no alternative. Either they came up with a catchy and unique piece of copy, or they wrote a straight script. Today,

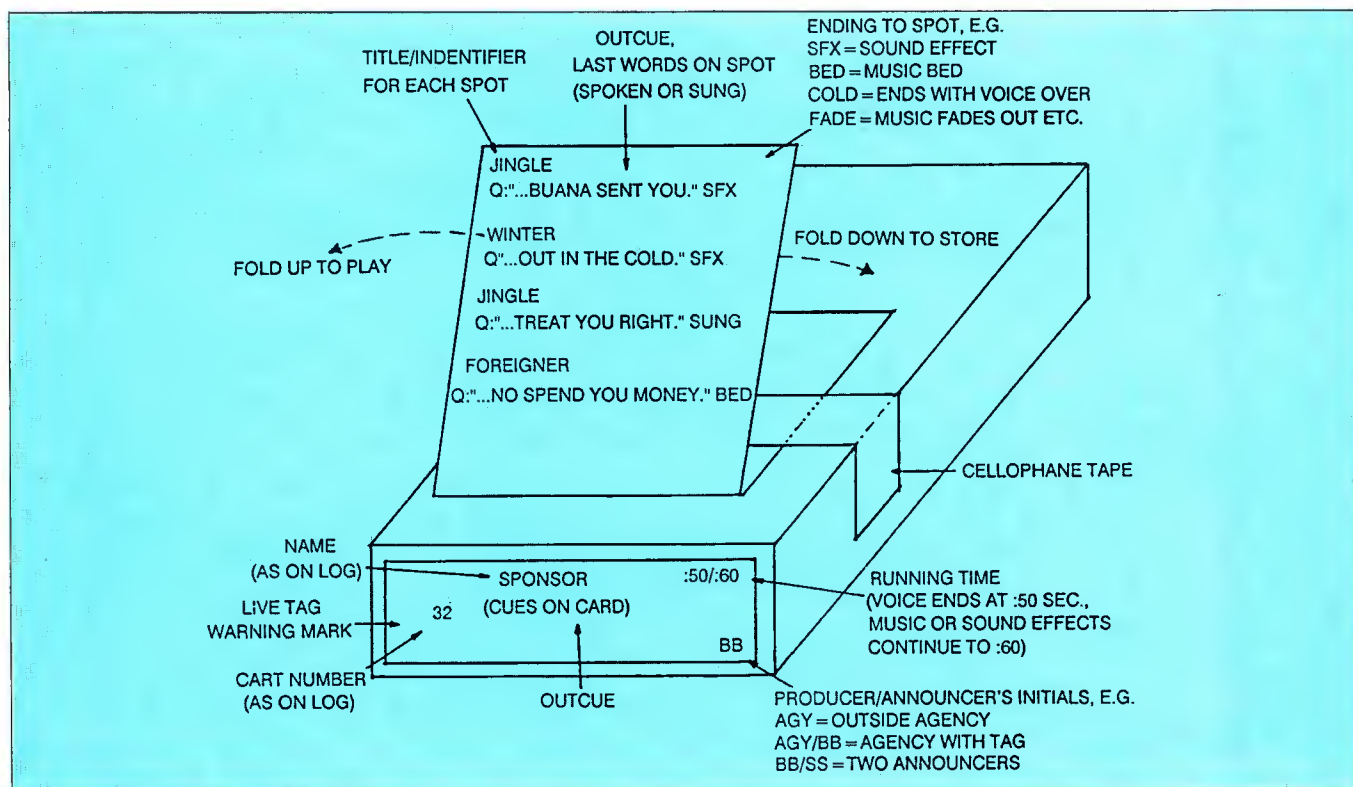


Figure 1. Identification of a cart.

not only can you sit down and type up an exceptional script, you can also embellish it with sound effects from CDs, bizarre processing and other electronic pyrotechnics. Either way, you can produce killer spots.

MAKE A CLEAN START

The simplest way to get rolling is by keeping what passes for a studio at your pleasedome in tip-top shape. Each day you must get on those tape decks with alcohol, cotton swabs and a head demagnetizer. And get a wastebasket and a vacuum cleaner. Come on, matey, swab them decks! Pick up some of the old coffee cups, tape labels and all those little paper disks the hole-puncher left behind. If you start with a clean studio (and keep it that way) you'll significantly cut down on the equipment failures and operating difficulties you face whenever you enter the studio. Grab a damp sponge or a handy portable sandblaster and scrub up those coffee cup rings and dump the overflowing ashtrays.

Now, how do you create? First, I want you to get some essential tools. You will need a good dictionary, a thesaurus, a typewriter (don't forget ribbon, paper and a bottle of Wite-Out) and some peace and quiet. Take the production order or client fact sheet and re-type all of the data yourself. This gets you focused on the material you'll have to write about. Now think about the sponsor's products or services. Who uses them? What is this particular merchant's competitive edge? (Price? Customer service? Quality? Name brand? Longevity? Variety?) Determine just one or two main selling points; the listener will only retain one or two main facts in an ad, so avoid the "laundry list" approach in which you run through the marvelous sale prices on two dozen items. The key is *differentiation*—you want people to be curious about this particular establishment.

You have but one goal: to bring customers to the client's doorstep. You do not have to sell anything. That's the client's job. You just bring 'em the prospects. So think it over. Does this spot lend itself to humor? Music? Fast moving edits and special effects? Straight narration? Try an interesting twist based on the sponsor's name (be it weird or extremely ordinary), the location, line of products or services or the method of selling. I've always liked creating unique characters that could appear in a series of commercials. Sometimes a theme can suggest a number of treatments. Comical possibilities include Great Moments in History, Bizarre Locations Around the World, Man in the Street Interviews, Weird Fictitious Employees/Products/Branch Locations and on and on.

TEN COPY TIPS

When you start to write the copy, keep a few specific rules in mind.

1. Open with a "grabber," an attention-getting lead.
2. Get the client's name into the first ten or fifteen seconds (people may tune out quickly).
3. Clearly describe whatever the sponsor sells. I've heard beautifully-produced ads that kept my attention, but left me wondering what the ad was *for*.
4. Repeat the phone number at least three times.
5. Make the location vivid. "At the corner of Guernsey and Trumbull, right under the water tower" is much easier to recall than "1376 Guernsey Street."
6. Create mental pictures. Your copy should give the listener a graphic image to remember. Go for adjectives and similes.
7. Get rid of unnecessary expressions such as "once again that number is" (simply repeat it), "And now an important message" (so, spit it out already), "so,

remember" (isn't that the point?), or "don't miss it" (brilliant comment). For years I bet copywriters that they could not show me an instance where they could not cut the word "located" from a script. So far I win.

8. Use *you language*. Talk to the listener. Address his or her needs. Why should they care? Make it a message that they'll want to hear. Uncover their motivation to take advantage of the offer. Is this particular commercial directed toward usefulness? Saving money? Status? Sex appeal? Fun? Uniqueness? Present it as a special opportunity.

9. Communicate one to one. Don't say "Attention, skiers," rather say, "If you ski..." Instead of "Here's a word for everybody who drives a luxury car," try "Driving a luxury car means you need..." I don't think of myself as a Taxpayer, but as a regular guy who happens to pay taxes. Talk to me as a person, not a member of some group.

10. Wrap it up with a powerful request for specific action. Tell the listener to stop in today on the way home from work. Have them call before Friday to get the special discount rate. Don't make the listener think. Nobody flips on the radio to see what the latest commercials are about. They're listening for entertainment. You have to turn them on.

Once you have written and produced a spot, the next step is to get it into the control room so it can be aired. Let's not drop the ball here. A mislabelled cartridge or a missing tag can damage the client's credibility, make the air personal-

ity look like a fool, or cost the station "make-goods." You are going to have to devise foolproof methods of assuring that the air staff plays your ad correctly every time. I always told the DJs where I worked that the ads were the most important part of our programming. Without the commercials, we were a terminal operation. No music or funny DJ talk was going to pay our salaries. A sloppy segue between tunes is a lot less harmful than a blown commercial. Get your perspective clear on this. With very few exceptions, any nitwit can be trained to spin records and spew out the time, temperature and a few witticisms. The trick is to blend that stuff with the spots to make it all work. So it isn't all that easy, after all. You can help, though.

**Because you can only air
one thing at a time, that one
thing had better be good.**

STATION OPERATION

Here are a few specific tips to keep those ads running smoothly at your radio station.

1. On commercials that require live tags, take a bright red felt tip marker and make a vertical bar on the left-hand end of the label. This prompts the jocks to look for a tag, even if the log doesn't. In addition, if you use music beds or "donuts" on cart, put a red bar on each end of the label. Two red marks should get anyone's attention.

Stereo echo, to be exact. There's also stereo chorus and flanging. Pitch change. Four kinds of reverb. Plus reverb and gate.

Thirteen different kinds of effects in all. In our new SPX90II, an encore performance of the most successful digital processor in audio history.

And now we've expanded the delay times. And expanded the possibilities.

There are 30 preset variations, each with up to nine separate controls. So you can get precisely the sounds you want.

But that's just the beginning. Because there's also room for 60 more custom variations, your own "signature" sounds that you can create and store in memory.

The SPX90II lets you label each custom effect with its own title. And you can instantly

There's an E



2. As long as you're fooling around with markers, experiment with high-lighter pens. I've effectively used yellow on station IDs, pink for public service announcements, green for station promos and so forth.

3. Eyeball each cart in your inventory when you first receive it and about once a week afterward. Look for loose parts, worn pressure pads, unseated guides and damaged tape. If you spot any of these faults, ditch the cart. It's usually not worth the time to repair a broken cartridge. Unless your station is hopelessly cheap, they won't mind tossing a dozen or two carts a month to be sure that no jam-ups occur on the air. Some stations even save them in a carton and when they get a sufficient number accumulated, ship 'em off to a recycling house to be professionally rebuilt and loaded with fresh tape.

4. Take brand new carts out of the carton and do the following:

Place a piece of Scotch Magic Transparent tape (3/4-inch width) across the area where the label goes, so you can easily remove them for re-labelling when the cartridge is reused for other things.

Tape often arrives fresh from the factory in a too-fresh, imperfect state. The oxide surface faces out and can collect microscopic dirt, dust, and may even have a couple of invisible clumps of excess coating. To polish it smooth, run them through a cart machine at least one full cycle. This removes any debris or unevenness from the surface of the tape. Now the cart will sound crisper and brighter from the very

first time it's used.

5. If a spot is to run for more than a week or so, cover the entire label with a piece of Scotch Magic Transparent tape (3/4-inch width). This keeps the label on even if it loses its stickum, and even prevents smudging by sweaty-palmed DJs.

6. Never record over a splice. On carts, check them visually, or better yet, use a mechanical splice finder machine to save time. I once worked at a station that had an amazing cart recorder—you put in a blank cart, it bulk-erased it, recorded a tone on the tape, then aligned the heads automatically to get the best signal, after which it re-bulked the tape and then shifted into a fast-wind splice search mode, and stopped just inches past the splice. In a total of about two minutes the cartridge was erased, aligned, tightened and set past the splice with the record light on and ready to roll. All the while I could be across the room selecting sound effects or music, or cuing the reel-to-reel. It spoiled me—and the thing wasn't unreasonably more expensive than most other decent cart recorders.

7. After you carefully bulk-erase a cartridge, run it through the cart machine for at least five seconds. This tightens up the tape. As a matter of fact, why not save time by checking for the splice after you've just erased the cart?

8. Always type all labels and live copy. A radio studio is no time to display your calligraphic talent. I don't want to hear you whining, "But I can't type!"

call up an effect with either our MFC1 MIDI foot controller, remote controller or just a standard footswitch (all optional).

But even if you don't need custom tailored sounds, the factory preset effects give you maximum signal processing in minimum rack space.

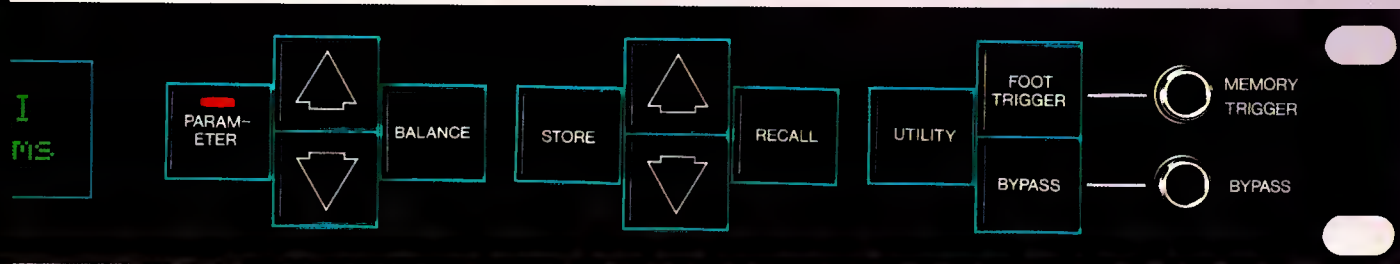
So whether you're a musician, producer or audio engineer, visit your nearest Yamaha

Professional Audio Products dealer to see and hear the new SPX90II.

It'll have some terrific effects on you. Yamaha Music Corporation, Professional Audio Division, P.O. Box 6600, Buena Park, CA 90622. In Canada, Yamaha Canada Music Ltd., 135 Milner Avenue, Scarborough, Ontario M1S 3R1.



cho in here.



Circle 14 on Reader Service Card



Figure 2. As an example of what this article discusses, we asked Long Island, NY's WBAB to photograph their control room. This is Production A. We thank WBAB's v.p. of programming, Bob Buchmann for the photo.

I'm a writer, and I'm finally up to pounding away at the keys with two fingers.

9. Always include start and stop dates on labels. Nothing sounds worse than a Labor Day Only sale promo running on September 12th. It makes you and your client sound hopelessly brain damaged.

10. Implement a consistent cart labelling format. My favorite is as follows:

TOP LINE—Name of sponsor in all capital letters, as it will appear on the log, and the precise running time. If the voice-over ends leaving a music bed, note both times, such as "52/:60."

SECOND LINE—Out cue (the final words on the ad). with a word that describes exactly how it ends, e.g. "cold," "fades," "sung," "spoken twice," etc.

SIGNAL—If your station indexes commercials by numbers, buy a few boxes of those little round colored dot stickers and type the spot number on one, then stick it to the far right or left side of the label. It's a lot neater than a scrawled "#042" somewhere on the label.

11. On cartridges that contain multiple rotating spots, use the folded index card technique (*Figure 1*). The card folds down for storage in the rack, and pops up in the operator's face while it's being played. Use all the extra space to be extra descriptive.

12. Use carts twice the length of the commercial to be dubbed onto it. (e.g., use a 2:20 cart and dub a :60 onto it twice with about ten seconds between.) This helps prolong the cartridge's lifespan, because the tape only passes completely through the machine once for every two airings.

13. If you put news actualities, drop-ins, IDs and so forth on a single cart, use a long one that gives you room to dub each one more than once in succession. This way you can check which one you've got cued in the audition channel. How does this work? If I have a cart with, say, nine different :05 second station IDs, I dub each one twice in a row. I can then check them in "cue" to be sure I know which one is going to play when I run it. If there are many cuts on one cartridge, consider dubbing each three times.

play (in cue) to time it and one play for air. Just remember to roll all three dubs before you put the cart away for next time.

14. You should never let any carts be pulled out of the player until they have reached the tone and stopped themselves. Replacing uncued carts in the rack will make you go to hell. If you need to yank out a spot that isn't re-cuing fast enough so that you can load the next one (usually only in emergencies—you should have more than two playback machines in your air studio), set it on the counter upside down to remind you to cue it after your break.

15. If you ever type copy to be read live in the on-air studio, pop it into an 8 1/2-inch by 11-inch plastic page protector. This keeps the type from getting smudged, and reduces paper shuffling noises if the announcer flips through the copy book when the mic is on.

COMMIT TO EXCELLENCE

One thing you have got going for you is that radio's frequency response and other audio specifications are in a unique range: better than most people can hear, and at least as good as the receiving sets and conditions they listen with. Nobody pops on the radio and expects Compact Disc fidelity (see accompanying sidebar). Consequently, you don't have to have nightmares about audiophiles picketing your front door with complaints of sonic mayhem.

In many of my jobs as a production director I had quite a few fierce confrontations with account executives, sales managers, general managers and owners who wanted me to air some of the most embarrassing and pitiful spots. They felt that the client's buck today outweighed the value of the station's image tomorrow. Maybe I was a bit idealistic, but it just always seemed as though there must be something we could do to at least improve a repulsive advertisement. Most of the time I won. Why? Because I built a reputation for producing superior spots of my own, because I demanded similarly high standards from my fellow workers, and because I didn't complain unless a spot was truly dreadful. I worked hard to establish a professional reputation and gain their respect for my critical judgement, and more often than not there was no sound logic to refute my claims. You have to keep your emotions out of it. Play for the team. Show that your concern is for the organization's ultimate profits, and that you are willing to cooperate and solve the problem in a mature manner. Napoleon Bonaparte instructed his subordinates to follow a principle which has become known as "completed staff work." This is a system by which a staff member who brings up a problem also has to submit two or three possible solutions along with a recommendation to choose a particular one. Now, obviously Napoleon didn't always take their advice, but at least they presented him with alternatives to look at. This saved time and mental torment, and usually resulted in great advances for the subordinates who reported to him thus. You can apply this method in your work. It transforms you from a crybaby into a leader. It shows concern for your station and it solidly builds your personal self-esteem.

Understand this: At any given moment, what is on the air at your station is your station. That includes all the production people, announcers, office staff, account executives,

management, even the equipment, building and office supplies. Everyone who works there, every tool you use, every image-building promotion campaign will be judged purely by the sound coming out of those speakers *now*. Your radio station's commercials are equally important as the jocks you hire and the music you play. Because you can only air one thing at a time, that one thing had better be good. If it's a commercial and you are in any way responsible, it should be something you are proud to hear.

I could go on, but that's what my regular column is for. Keep up with me as I toss out more advice and commentary, and send me a note when you get a chance. If we broadcast professionals and recording producers stick together and share our knowledge, everybody gets to contribute to a stronger personal and global economy.

TALKBACK MIC

Thank you, Allen D. Schultz of ListenUp in Denver for technical help...Marc Putman of Esconitas, CA sent in a killer demo tape...Also thanks to Tom Norton of Kinetics West for the assistance with our new audio booth at CareerTrack...I just listened to a cassette from Sam Beniquez of Albany, NY that tells me who the hottest new ad producer is in the capital of the Empire State...The Chief Engineer at KHX/KTCL, Jim Mross, Jr. has my vote for *Ad Ventures* fan of the month...Ritchie Kay and Fran Dresser have a promising new jingle company now open for business in that semi-toddlin' town of Lincolnwood, IL...Once again, Greg Diaz of Tascam comes through for me...Nutmeggers, check out the superior customer service and facilities at the CourtHouse health clubs throughout Connecticut, managed by Mr. Gary Rogers, soon to be hitched...Open memo to AES officers: how about a conference somewhere in the middle of the country? (Editor's note, an AES Conference is coming up about now in Nashville, TN.) I can't always afford Paris, NY or even LA...A long-overdue Hello to John "Garlic" Barilla...BEST NEWS OF THE YEAR SO FAR — Gumby is coming back to TV this year!

SHAMELESS SELF-PROMOTION DEPT.

The ability to create successful radio commercials can open doors to major career advances and bring you substantial additional income.

My latest project is pretty exciting (to me, anyway). I'm producing a comprehensive audio-cassette album on how to produce radio commercials. It will be a combination of ideas, hints, tips and many interesting samples of my own best commercials, including some that brought me recognition with various awards including the Clios, the International Radio Festival of New York, the Long Island Advertising Club's Best of Long Island, the Big Apple Awards, the New York Metropolitan Radio awards and more. I'll describe in detail how I put them together, and give you plenty of suggestions on how you can write and produce better spots. This program will be useful for radio station personnel as well as operators of independent recording studios.

It's planned for release in mid 1988, and the price is still pending. To be sure you receive an opportunity to get your copy of this exclusive tape set, please drop me a note at the following address:

Brian Battles
c/o CareerTrack Publications, Inc.

1775 38th St
Boulder, CO 80301

Your inquiry places you under no obligation; I'll simply send you a letter when the product is ready.

WHY CDs?

Regular readers of my *Ad Ventures* column are aware of my feelings toward the insignificance of playing compact discs on the radio. Compared to conventional LP records, CDs offer a few advantages and a number of notable non-advantages, so to speak.

The advantages:

1. Storage space — CDs are smaller than records.
2. Ease of use — They're quick and simple to cue.
3. Durability — They don't wear out or get damaged easily.
4. Quietness — They contain negligible background noise.
5. Content — CD versions of albums often contain an extra song or two that won't fit on a record.

The non-advantages:

1. Frequency response — Conventional records already exceed radio stations' technical limits.
2. Wider dynamic range — Rather meaningless since stations just compress the signal anyway.
3. Some older recordings actually sound *worse* on compact disc since the added clarity makes defects and inferior recordings more prominent.
4. They don't skip (although I've heard countless episodes of bizarre electronic pandemonium when a compact disc "glitches out"). Actually, the issue of durability is dubious, since most stations get all the records they want for free, and if they do have to pay for an album, a CD costs a lot more than a record.
5. The liner notes are often sketchy, cumbersome or just too small to read. The cover art is like a Picasso on a postage stamp. But this may not matter if all you care about is what the audience hears.
6. Then there are times when you may wish to vary the speed (pitch); with compact discs it's usually not possible.

Are they handy to cue? Yes and no. If all you do is pop 'em in the player, select a track and press "START," then the answer is yes. If you prefer the feel of slip-cuing a platter by hand, or if you ever want to try stuff like flanging, disco-style live splicing or any other creative effects, you're pretty much out of luck. Consider the issue of consistency: If you can truly hear the difference, what are you going to do about playing a mixture of fresh CDs and albums available only on vinyl? There are plenty of old LPs and 45s out there that aren't available on compact disc (though admittedly, there's more coming out every day).

So what's the verdict? Mainly, that CDs are terrific if you have a high-quality player and never try to fool around with the sound. Records afford more flexibility of use. We can call the CD the "lazy DJ's" format. It depends on your station's style. Forget the technical improvements; by the time your signal gets blared out of somebody's boom box or average stereo system, there is almost no audible difference. Weigh the factors above and make your own decision.

db

Broadcast Audio

THE ORIGIN OF THE VU METER

● How did the now-ubiquitous vu meter come to be adopted as the standard audio level monitoring device in the United States? The advent of broadcasting and networking immediately produced the requirement to monitor audio levels.

Program audio level is measured with some form of indicator possessing particular static and dynamic characteristics that determine how it reacts to audio program energy. The standard volume indicator or vu meter has historically been the audio level measurement device of choice among broadcasters in the United States, while in much of Europe the peak-program meter has occupied this niche. There are numerous and often spirited debates between proponents of these two types of audio meter, but both types have been successfully used since the 1930s for the measurement and supervision of audio program levels. Let's explore the origin of volume indicators and the subsequent development of the U.S. standard vu meter.

AC theory teaches that there are three related values by which the magnitude of a sine wave or any other periodic waveform may be expressed: the average value, the rms value, and the peak value. When audio program material is addressed, waveforms are encountered which are often quite complex and usually not periodic. The magnitude of such waveforms cannot be expressed in simple numerical terms of average, rms, or

peak values, because not only are these values constantly changing, but they also appear to be affected by the characteristics of the device used to measure them.

THE FIRST VOLUME INDICATOR

The first volume indicator was developed in 1921 upon the occasion of the burial ceremonies for the Unknown Soldier. The ceremonies were to be heard on public address systems by audiences in Arlington, New York and San Francisco. Some preliminary tests had revealed that amplifier overload distortion was more objectionable when heard from a loudspeaker than from a telephone receiver. To avoid overloading the telephone repeaters used in this early "network," a device was developed which gave a visual indication when speech level was high enough to cause repeater overload.

This led to the development of a measurement apparatus consisting of a triode detector connected to a milliammeter, with a potentiometer on the input to adjust sensitivity. Reference level was chosen as that level of speech which, when transmitted on the telephone circuits in use, would cause the repeaters to be just on the verge of audible distortion.

...this created a situation
of great confusion...

The birth and early growth of broadcasting and radio networking produced a plethora of different volume indicators used by different organizations. These volume indicators were both average- and peak-reading

types having slow, medium or fast pointer speeds; half- or full-wave rectification; critically to lightly damped movements; and reference levels based on calibration with 10^{-9} , 1, 6, 10, 12.5 or 50 milliwatts in 500 or 600 ohms. This created a situation of great confusion, particularly when an effort was made to correlate the readings of one group with those of another.

In January, 1938, the Bell Telephone Laboratories, the Columbia Broadcasting System and the National Broadcasting Company began a joint development effort intended to generate a uniform practice for measuring volume level. The fruit of this labor was a new volume indicator, a new reference level, and a new terminology for expressing measurements of volume level. These new volume measurement concepts were adopted by CBS, NBC, and the Bell System on May 1, 1939, and to this day they remain in virtually universal use in the United States.

PEAK VERSUS AVERAGE

The first important decision to be made was whether the new indicator should be of the peak-reading or the average-reading type. The peak-reading indicator was in common use in Europe by this time, and was used to monitor transmitter modulation in the United States, while U.S. broadcasting networks and general telephone use favored the average-reading type of indicator.

An average-reading meter consists of some form of rectification and a d.c. milliammeter, and is somewhat slow to react, generally requiring tenths of a second after application of a signal to reach full deflection. This

type of indicator integrates whole syllables or words. A peak-reading meter would be more accurately called a quasi-peak meter, because it does not read true instantaneous peaks, but rather has an integration time in the milliseconds range, thereby averaging several peaks of audio program material. The difference between the two types of indicators is principally one of degree.

The volume indicator working group felt that the three most important uses for the instrument were the indication of a suitable level of audio program material to avoid audible distortion, checking transmission losses or gains in an extended network by simultaneous measurement of the same program peaks at a number of points, and indication of the comparative loudness of programs when converted to sound.

Extensive listening tests were undertaken using the judgement of observers as to when distortion could be heard. For the purpose of these tests, the best volume indicator was considered to be the one which most nearly approached the same reading on all the test programs when overloading could just be detected. The program sources included direct microphone pickup of speakers and musical instruments, as well as high quality recordings. Two different types of amplifier were used in these tests, one having a sharp cutoff characteristic, the other a more gradual cutoff. It was concluded that for the sharp-cutoff amplifier the peak meter permitted about 1 dB higher average level to be transmitted for the same likelihood of audible distortion than did the average-reading meter. For the gradual-cutoff amplifier, no advantage was detected for either type of meter in the indication

The second stated use for a volume indicator, simultaneous checking of specific peaks at many points along a program network, was important because this was the only way to check gains on such networks during the

broadcast day. For this function, the averaging meter proved unquestionably superior to the peak meter. Phase distortions and nonlinearities present on program networks, while not audible, changed the shapes of

Found, The Missing Link. Gauss Coaxial Monitors

It's a well known fact that loudspeakers are the missing link in studio, post production and broadcast facilities' audio chain. The accepted criteria for ideal speakers are: balanced, phase-coherent or time aligned, and with as little color as possible.

Gauss Coaxial Monitors let you hear it all, even the mistakes... without adding color. These time coherent monitors provide an extremely stable stereo image so you know exactly what you're mixing. And, if you're mixing digital sound, they offer the cleanest reproduction you've ever heard... with no high-end harshness. And, with 400 watts of power handling, you'll hear all the dynamics.

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**The one-milliwatt
calibration was selected
because it is a simple,
round number...**

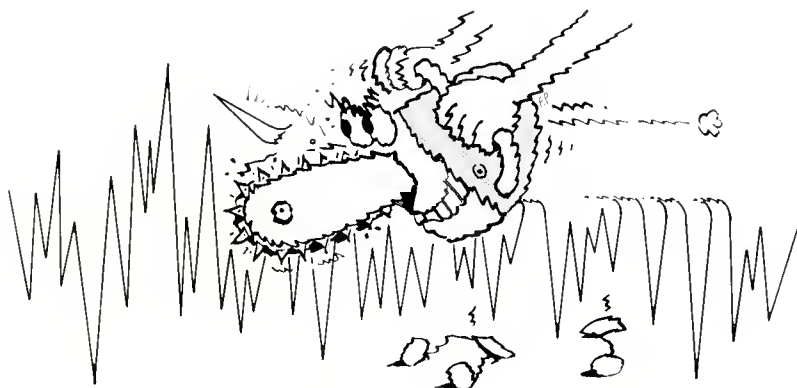
of audible distortion. It was concluded that use of an averaging type of meter offered little or no disadvantage in the detection of audible distortion, and it was felt that this was so because the human ear fails to notice considerable distortion of rarely occurring peaks of short duration.

Circle 16 on Reader Service Card

program waveform peaks, causing serious errors in readings obtained using peak-reading meters. Average-reading meters with their longer integration times were immune to this effect. To test the correlation between

volume indicator readings and the perceived loudness of program material, groups of listeners were asked to judge the loudness of programs and these opinions were compared with meter readings. Extensive tests were

conducted with material that included male and female speech, dance orchestra, brass band and piano music. The loudness tests demonstrated no significant advantage for either type of instrument.



DOES YOUR LIMITER MASSACRE YOUR SOUND?

The Apex Dominator™ is the perfect solution!

Unlike dumb, over-threshold devices, the Dominator is an intelligent 3-band limiter with a proprietary circuit which varies the threshold for limiting. The result is an *absolute* peak ceiling while retaining a transparent sound. You can run hotter levels to maximize signal-to-noise without fear of overloading.

The Dominator provides total transparency below processing threshold...increased loudness...freedom from spectral gain intermodulation...maintenance of transient feel...high density capability...and can be used for multiple applications. It's flexible and easy to use.

Ask your audio professional for a free demonstration. Once you've heard it, you'll never be satisfied with your old limiters.



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CONCLUSIONS

Audible distortion tests demonstrated a slight advantage for the peak-reading meter, while peak-checking tests indicated a marked advantage for the average-reading meter, and loudness tests showed neither meter superior for that function. The averaging meter could include a copper-oxide rectifier within its case, but the peak meter required a tube circuit and its power supply. In addition to its program peak-checking superiority, the averaging meter offered the advantages of relatively low cost, ruggedness and portability. It was therefore decided to develop the averaging meter, and thus the standard volume indicator or vu meter was born.

BALLISTICS

Extensive conversations with operators and others intimately involved with the use of volume indicators produced the conclusion that for ease of reading and avoidance of eye fatigue, the meter movement should not be too fast. The new standard specified that sudden application of a 1000 cycle (yes, but it's 1939, remember?) sine wave of an amplitude which would cause steady-state deflection to the reference mark would cause the pointer to reach 99 percent of its final deflection in 0.3 second. It was also determined that to avoid a jittery action the movement should be slightly less than critically damped, so that sudden application of the previously specified sine wave results in overshoot of 1 to 1.5 percent.

A small but important aspect of the scale was the inclusion of an arc under the scale numbers to reduce eye strain...

SCALE

The scale was developed after polling many operators. It reflected their preferences for an orange-yellow

Circle 17 on Reader Service Card

face to reduce eye strain, the use of about 71 percent of scale length to the reference mark rather than the then-traditional half-way point, a spade rather than lance type pointer, and general agreement that a 3 dB indication above the reference mark was adequate. A small but important aspect of the scale was the inclusion of an arc under the scale numbers to reduce eye strain by giving the eye a path to follow, rather than having the numbers "float in space."

REFERENCE VOLUME LEVEL

The new reference volume was chosen to correspond to the "0" reading of the new volume indicator when calibrated with one milliwatt when the meter is connected across 600 ohms. The one-milliwatt calibration


The new term "vu" was introduced to express the units measured by the new indicator.

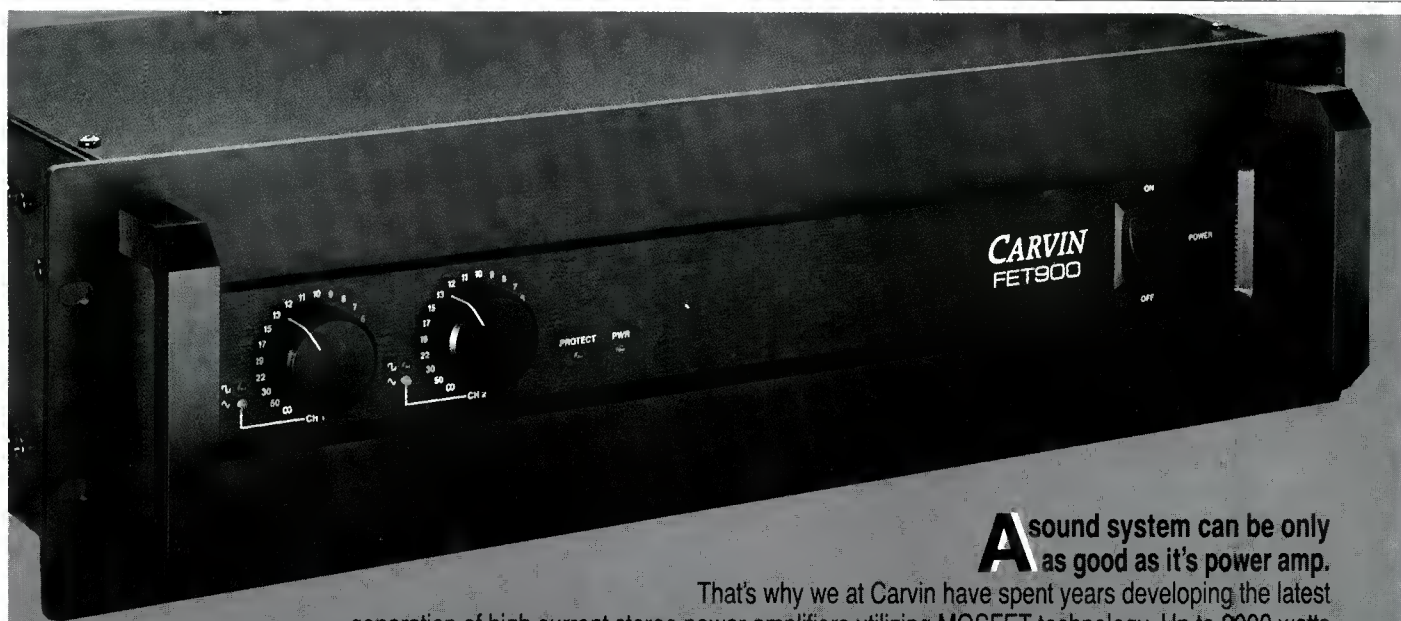
was selected because it is a simple, round number and a level that was often used in telephone plant testing. Calibration was based upon a power rather than a voltage to avoid apparent gains or losses at points of impedance transformation. The standard impedance was chosen to be 600 ohms because there was more 600 ohm equipment in use than any other impedance.

The new term "vu" was introduced to express the units measured by the new indicator. The number of vu was defined to be numerically equivalent to the number of decibels above or below the new reference volume level. It was intended that when one

encountered a measurement in vu, it would be understood that the measurement was made with the new volume indicator and with respect to its new reference volume level.

When the history of the vu meter's development is known, the reasons for its selection as a standard are apparent. Although these reasons may be less compelling in 1988 than in 1938, the vu meter has stood the test of time, nearing the end of its fifth decade as the standard volume indicator used in the United States broadcasting.

Before the vu meter was developed in the United States, the peak-program meter had become the instrument of choice in Europe. Next time, we will take a comparative look at the two devices. 



A sound system can be only as good as its power amp.

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Circle 18 on Reader Service Card

Digital Radio: The Future Is Now

Digital radio is a reality for a contingent of Boston audiophiles. As discussions escalate over the allocation of spectrum space for High Definition television, mobile land communications and digital radio, Boston's public broadcaster WGBH is regularly transmitting a PCM video format digital audio signal to several hundred listeners in eastern and central Massachusetts.

CITING THE GENERAL PUBLIC'S ACCEPTANCE OF compact discs and the potential of digital audio transmissions to provide a signal with lower noise, wider dynamic range, lower distortion and a more uniform frequency response than conventional AM or FM broadcasting, the Federal Communications Commission granted WGBH a temporary experimental authorization to transmit on WGBX-TV, UHF Channel 44, a PCM encoded digital signal during periods when the station is not broadcasting its normal television programming. Utilizing a Sony PCM 501 digital audio processor to encode stereo audio information into video format digital audio, WGBH is transmitting a PCM signal in the portion of the television signal normally used for the transmission of picture information. When a conventional television receiver is tuned to WGBX during the digital audio broadcasts, viewers see a modulating pattern of black and white dots representative of the digital information. In addition, a monaural reference of the digital audio signal is transmitted on the normal TV audio channel supplemented with announcements describing the nature of the broadcasts.

WGBH is transmitting a PCM signal in the portion of the television signal normally used for the transmission of picture information.

DIRECT DIGITAL BROADCASTING

To date, most of the digital programming has been a simultaneous broadcast of WGBH Radio's programming. In recent months, the broadcast schedule has included all compact disc programs, digital recordings made in Boston

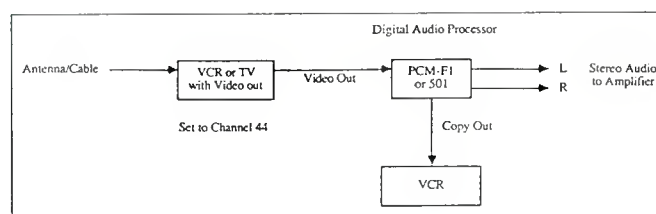
area concert halls and live in-studio performances by both noted area musicians and renowned classical artists appearing on WGBH-FM's perennially popular morning show, Morning Pro Musica.

The most exciting events in the digital broadcast's short history include live transmissions of orchestral concerts from Vienna and Tokyo. In May, 1987, WGBH-FM, in cooperation with FM Tokyo Broadcasting, presented a live broadcast of the New Japan Philharmonic under the direction of Seiji Ozawa and featuring soloist Kathleen Battle from Tokyo's Bunka Kai-Kan. This broadcast was live on both WGBH-FM and WGBX-TV and delayed for national distribution via the public broadcasting satellite system. As engineers in Boston received the live Trans-Pacific feed of a PCM signal, they routed it to WGBX for broadcast.

WGBH views its current experiments as achievements which may have the greatest impact on the future of broadcasting.

Listeners to the digital service were treated to a historic first, a live concert broadcast which had traveled in the digital domain from Tokyo via two satellites to their homes. As one listener remarked, "The sound was spectacular, capturing the range of the music and the ambience of the concert hall." On New Year's Day 1988, WGBH repeated its

Figure 1. How digital audio transmissions on channel 44 can be received and recorded.



John Voci is the Operations Director of WGBH-FM. He is an occasional contributor to db on digital audio.

earlier triumph by routing an incoming feed of the Vienna Philharmonic's annual New Year's concert directly to listeners. This live broadcast of Austrian Radio's feed enabled WGBX listeners to experience the Vienna feed concurrent with the engineers and producers in the WGBH studios. Again, the broadcast met with enthusiastic response from the several hundred audiophiles who regularly tune to WGBX for the digital service.

The 90 dB of dynamic range and signal-to-noise... far exceeds the performance of the best FM transmitters...

THE LISTENING PROCESS

To receive the digital broadcasts, listeners need either a TV with video output ports or a VCR tuned to WGBX and a PCM processor compatible to the Sony 501 (Sony F-1, 601, 701 or models manufactured by Aiwa, Technics and Sansui). The video signal is decoded by the processor, converted to analog and routed to an amplifier for playback. By using a VCR, listeners have the option of time shifting the digital broadcasts for playback at their leisure.

For the engineering management of WGBH, comparing the digital broadcasts to conventional FM is analogous to comparing CDs to vinyl discs.

The 90 dB of dynamic range and signal-to-noise in conjunction with the flat frequency response of the video formatted digital audio far exceeds the performance of the best FM transmitters and is capable of delivering to the consumer audio comparable to compact discs.

The FCC's continued support of the WGBX experiment indicates their interest in the potential of digital broadcasting.

Can a Monster Cable really make a difference?

Here are a few people who believe it can.

"We now use Monster on every project to the extent that we would not consider making a recording without them. We've flown Monster Cable all over the world to achieve that goal!"

— Jack Renner, The Telarc Digital Label, Cleveland

"If I had one wish, I'd wire every tape machine, every monitoring system, every console — in fact, every recording studio I've ever worked in — with Monster."

— John Arrias, Recording Engineer/Producer, Los Angeles

"It's the only way I can maintain a reference to accurately record, playback, and transfer what is on the tape."

— Ian Eales, Recording Engineer, Los Angeles

"I insist on Monster for all my recordings. It lets me capture all the sound that's missing with other cables."

— Jeff Balding, Recording Engineer, Nashville

"In my 20 years of building recording studios, all the amps, consoles, recorders, loudspeakers — everything I've run across, combined — has not made the difference Monster Cable's wire technology has."

— Ed Bannon, TAJ Soundworks, Los Angeles

"Due to Monster's 'phase-alignment' technology, it was like a mask, a veil, had been lifted from the sound."

— Bob Hodas, Recording/Concert Engineer, Sausalito, CA

"I can't believe that all this time I've been EQing for my cables! Now I'm getting so much sound I find myself using much less EQ."

— Randy Kling, Mastering Engineer, Disc Mastering, Nashville

"It was a little frightening, the difference we heard with Monster Cable. Suddenly the stereo image was better, the tightness of the sound was better, the openness was better."

— Bob Ludwig, Mastering Engineer, Masterdisk, New York

Something's happening here. But this time, it's exactly clear.

At least to the growing number of audio professionals in recording studios, mastering rooms, and feature film sound effects facilities.

They've discovered the significant performance differences Monster makes in their work. And they consider Monster Cable to be a milestone achievement in audio engineering.

They're pioneers. But they were once skeptics. Until they opened their minds to the idea of high-performance cable. And their ears to the sound of Monster Cable.

Now some of them won't even work without Monster.

Must be because of Monster's innovative cable technologies and construction.

Like "Bandwidth Balanced®" multiple-gauge wire networks, "MicroFiber™" dielectric, and "Duraflex®" jacketing.

Each an advanced technology other cable manufacturers can only dream about.

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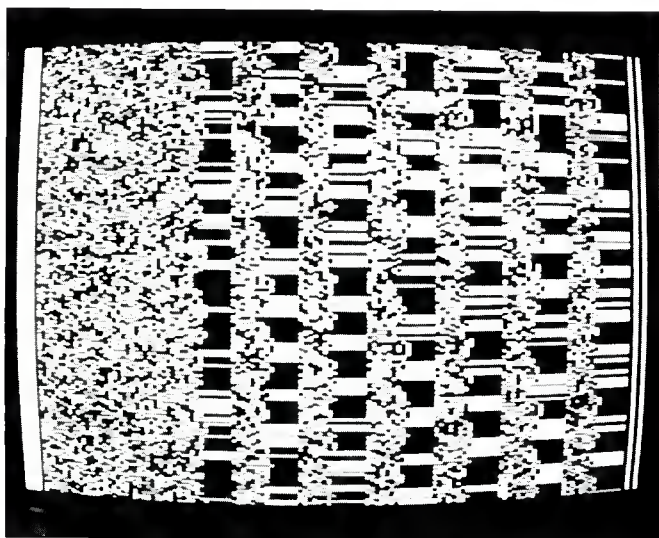


Figure 2. A video monitor showing the PCM digital audio signal of WGBH Radio transmission, as it appears on WGBX/Channel 44. Photo credit: Cobern Bennett/WGBH.

NEED FOR SPACE

WGBH's recent experiments with digital audio is the most current chapter in seven years of exploring both various digital formats and methods of transmission. Awarded a 1985 Edwin Armstrong Award for Technical Achievement for the first point-to-multipoint digital audio broadcast (db, November/December 1985),

Japan has already taken the lead in planning for a digital audio service.

WGBH views its current experiments as achievements which may have the greatest impact on the future of broadcasting. With the entire audio industry in the midst of a revolution that is transforming how both professionals and consumers view high fidelity audio, only broadcast lags in converting from analog to digital. Since the advent of stereo FM, no major technical innovations have occurred in radio broadcasting in the past quarter of a century and of the three major audio mediums—disc, tape and radio—two have already converted to digital. For David MacCarn, WGBH's director of engineering and the impetus behind the WGBX broadcasts, it is imperative that broadcasters, equipment manufacturers and the trade press examine the future viability of a digital audio service and advocate for its consideration in the allocation of spectrum space. "If we don't raise the issue now," states MacCarn, "it may be too late in a few years or we may see digital audio allocated spectrum which is impractical to the general consumer."

Japan has already taken the lead in planning for a digital audio service. Utilizing direct broadcast satellite and a PCM signal with a 32 kHz sample rate, a standard that can be decoded by R-DAT recorders, Japan is poised to introduce a digital service in 1988.

WGBH engineers concur that there is an avid interest by many for a high quality audio service.

Digital formats and broadcast standards aside, after sixteen months of experimentation, WGBH engineers concur that there is an avid interest by many for a high quality audio service. The beauty of the WGBX broadcasts is that it uses existing technology which, although not widespread, has already gained acceptance by professionals and a small segment of the consumer market.

LIMITATIONS OF EIAJ

The EIAJ video format digital signal is not without its drawbacks. Because the Sony PCM signal presently utilizes approximately 2.5 MHz of the 6 MHz of bandwidth allocated for a conventional television signal, MacCarn acknowledges that the present 44.1 kHz PCM video format digital audio signal may use too much bandwidth for widespread acceptance. "There are alternatives," concedes MacCarn, "it's conceivable that an enhanced audio service could use 1 MHz of bandwidth or that a system similar to the Dolby Adaptive Delta Modulation system which fits twelve stereo channels into 6 MHz of bandwidth may be sufficient." Aside from improved audio fidelity, MacCarn views digital and the error correction schemes inherent in the medium as having an application in decreasing multipath interference, a problem experienced by FM broadcasters in urban areas.

The FCC's continued support of the WGBX experiment indicates their interest in the potential of digital broadcasting. Information gathered from the WGBX test will be useful in determining both the technical viability of transmitting digital audio and the general interest in a digital broadcast service. Naturally, general acceptance and the cost benefits of a new radio station are questions which remain unanswered. Presently, WGBH is seeking new methods to test and is willing to cooperate with other manufacturers and broadcasters on new technologies which may ultimately prove successful in introducing digital radio to the consumer. Whatever the future brings to the broadcast industry, for WGBH and its listeners the future of digital radio is now. db

...WGBH is seeking new methods to test and is willing to cooperate with other manufacturers and broadcasters on new technologies...

The Mother Ship Revisited

SINCE THEIR MOVE FROM Franklin Square, N.Y. into the Kaufman Astoria Studios, Master Sound Astoria's owner/chief-engineer Ben Rizzi and partner Maxine Chreйн have been continually analyzing, building and expanding MSA. The move to the Kaufman Astoria Studios, in 1985 (cover: *db Magazine*, September/October 1985), took a year of planning and more than three million dollars to realize. On March 4, 1987 at MSA, *db* covered a bi-coastal live recording session via satellite—an event that established MSA as a “mother ship” for future sessions of this nature.

In the recording industry we have noticed a trend. Many of the “world-class” studios have abandoned album-oriented clients in an attempt to establish a more lucrative dollar-per-hour post-production clientele.

Not so at Master Sound Astoria (MSA). This studio has possibly found the answer to a very complex problem—how to maintain a balance between the light speed paced world of post-production and recording artists who often require huge blocks of time.

THE RIGHT TOOLS FOR THE RIGHT JOBS

Ben Rizzi tells us, “Post-production is a relatively new endeavor for Master Sound. Before we got into *post*, we researched it very carefully. The more I researched, the more I found that there were points of weakness when it came to a total understanding of audio for T.V. and its potential impact. Many try to use audio like a multi-track. For instance, in a given sit-com there might be one track for dialogue, one track of audience and mix it mono.”

Corey Davidson is the Technical Editor of this publication.

What Ben is describing is basically what has been the case in the movie industry, where dialogue is almost always kept in the center channel and any other audio information is incidentally along the side(s) of the center channel.

Ben continues, “What’s interesting here is that we are involved with both the film technologies and the televi-

be a tough job over here because of the combination of film and television. One of the first installations was the Telecine setup, a device which transfers film to video. Normally, we receive a reel of picture and a reel of sound. We put the picture on a holographic projector or a Telecine machine and we put the sound on a separate dubber. Then the two are interlocked and a transfer is made to video. The problems that arise from getting into the post ball of wax is that the technical upkeep poses new problems that give it the character of an entirely separate business. Having a telecine means that every month there are new types of stabilizers that we must be aware of, not to mention the various devices designed to improve film motion. It never ends. Every week we are updating our equipment.”



Figure 1. David Browning in Studio A2s control room. David is MSA's chief audio post engineer.

sion technologies. We are doing films and television, so the first thing that we had to decide upon was whether a film-chain should be the first priority. However, we have a unique situation here in the sense that we never know from day to day what productions are coming in. Our job here is to cover the audio for all of the various projects, shows and movies that come in. For a long time we were doing a bang-up job with music recording, but somehow, I found many of the post-production clients going to Manhattan or back to California to finalize their productions. Our decision to go post was well cemented when we realized that there was business that could be had if we went post and really did it right. We hired a handful of some of the finest post-production people and had meeting after meeting about what it would take to become a major post facility. The more we talked about it, the more we realized that it was going to

A NEW APPROACH TO ADR

How many tracks can actually be put onto video tape? Ben answers, “We can only put two tracks of audio onto the video tape. We use those two tracks primarily for the *rushes*. These are the work materials (versions of the productions) that get sent back and forth between the lab and this studio. We have a screening theater here, but we might need to send copies of the rushes to California, the producer, the director, and/or the script writers so that they can look at the daily productions and see what might have to be changed. It is typical for us to have to run-off half a dozen three-quarter and half-inch video cassettes on a daily basis. By having the Telecine capability we no longer need a projector or a projectionist when we do scoring because we can put the film up on the Telecine machine and the machine generates SMPTE time code. What we then do is have the digital recorders “look at”

the SMPTE time code. This enables the digital recorders to *chase* the film chain. The same material can be projected from a video projector onto a twenty eight-foot, diagonally measured screen. Finally there is a matrix of video monitors throughout the complex so that any of the rooms can "look at" any of the films.

The important aspect is that the Telecine machine opened us up to other areas of post-production. Most recording studios' post-production capabilities are in the realm of video only. MSA can now transfer the film and lock it in with video which enables us to do ADR (audio-dialogue replacement) in a new way. Currently when people do ADR, they do it on film. A loop of ADR is put up on a projector along with a loop of the sound. This projection technique is faster than the normal speed. Then they use a device that gives them three beeps. At the end of the third beep they can replace the dialogue that's on there because there's a third sound recorder locked into the film chain. All these steps make this kind of editing a very cumbersome process because you do one scene, take all of the loops off, spool up and do another scene, etc. Now, with the Telecine, we have a way of transferring picture with sound in sync and generating time code to any kind of video tape that we want whether it is Beta or half-inch or three quarter-inch or even one-inch."

Now we can instantly give clients any configuration up to twenty four tracks.

Ben has thoroughly described the primary formatting abilities of his studio. He continues to explain to us what is involved in completing a studio where a client can work faster and more easily than in most conventional post-production facilities, "The next logical approach was to establish state of the art ADR. We purchased the Adams Smith 2600 controller with synchronizers." David Browning head of post production says, "Now we can instantly give clients any configuration up to twenty four tracks." Ben states, "As they overdub, they can do one, two, three, four...up to twenty four overdubs because we have taken a video machine and locked it in with a twenty four-track recorder, a two-

track recorder that's running constantly with Dolby keeping every take, plus a film machine that's locked into that chain constantly recording the last take only. The ADR people thought that they were spoiled having worked with film, thus having the ability to record up to four different takes on a four stripe. Now they can have any amount and configuration of tracks that are analog or digital." David adds, "Our clients don't have to change the way they work, however, we have the capability to greatly increase their working speed."

COMPUTERIZED EDITING SYSTEM

Another outstanding feature is that this is all totally computerized because the 2600 is a computer. At the end of the day the client's edit decision lists and all of their comments on each and every take is in this computer, besides the fact that the computer generates a CMX edit decision list. This kind of processing opens up other avenues because super state-of-the-art film equipment is materializing that utilizes a similar edit deci-



Figure 2. This photo offers a perspective on the size of Studio A1. One hundred thirty-five kids at a songfest.

sion list. If it's a television show and the ADR is going to the post place, we can either read their edit decision list, have it come up and find out where they have their problems, or we can send them our edit decision list and they'll know exactly where every drop-in is for that ADR. What we are also trying to do with all of this super technology is to swing our clients over to digital. Slowly but surely that's starting to happen. Due to the fact that we are so heavily into video post-production, we had an easy time deciding upon which video

machine we should have...a Sony BVH 2800 with digital audio. There aren't many of those machines around but they're starting to appear out there in the market place. What we normally do for every client that comes in here (such as National Geographic, Showtime and various film and video people) is give them a choice of audio formats. So far our experience has been that these clients have been impressed by the sonic quality of their audio when recorded digitally, not to mention the superior editing capabilities. Some clients do not have the digital capability, however, if they ever have access to digital down the road, digital is part of their sessions here so they have it on hand as well as the analog audio tracks."

What has the traffic been bringing you as of late? Ben replies, "Lately, much of the work has been scoring. What we have been discovering is that most of the composers and contractors are used to doing music scoring. They really know about separation. However, we have been able to supplement their knowledge by bringing to their projects an en-

hanced understanding of ambient rooms, acoustics, and what it takes to get an improved sound. So what we have tried to do is to show them a way of working that is very fast and the improvement of sonic quality, while preserving the separation that they want."

THE SOUND OF SIT-COMS

What has been the orchestral size of these projects? "The scoring has been anywhere from seven or eight pieces to fifty five pieces."

Are you set up to do scoring from purely electronic instruments at this point? "Yes, as a matter of fact we have a Fairlight now. We are going to be one of the first facilities in New York to actually mix sit-coms." What is so special about sit-coms? "Number one, there is only a handful of *laugh* people and most of them are based in California. Up until now they have had a monopoly on a laugh machine. This machine used by the laugh people is a little black box consisting of nothing more than endless loop cassettes or tapes. With the touch of a button and a couple of

faders, more or less laugh is played. Number two is the fact that these laugh people are using a machine that does it exclusively for this select group of people. To give them a sampler and a keyboard, they're lost. So what we've done is develop our own black box interface that will work with various samplers. The person that is comfortable with the old button and fader format can work that way. The person that is comfortable with a keyboard can work that way and the person who wants a touch-sensitive set of conditions can have that too. More important than the equipment is the talent to do the job. We have that talent here. We have joined forces with Sit-Com Services, people who are ultimately experienced with the format and "sound" of sit-coms. One thing that we have tried to do is to take advantage of all the digital equipment that we have and use it to an advantage."

KEEPING IT IN HOUSE

In one way Master Sound is an independent recording studio. In another way it's actually a part of the Kaufman Astoria complex and Kaufman Astoria is doing a lot of film and T.V. shows. So where is your business coming from? Is it coming down the wire from what is being shot here anyway, giving them everything under one roof, or is it coming in just for you from somewhere else? "Of course we're getting work from Kaufman Astoria but about seventy percent is outside work. The sit-com work is all outside work. It's nice to have people working here and all these wonderful productions going on. Occasionally they come down and ask for, say, a list of thirty special effects. Our people then assemble that list for them. But most of our work comes from the outside and it's very diversified. Another interesting observation that can be made is that there are studios in town that specialize in rock and roll, and that's all they do. Then there are studios that specialize in jingles, and other places that specialize in acoustic recording (like an orchestra). We, fortunately, wear many different hats. The success that we have had with rock and roll can probably be attributed to the sound of our 'big room,' our pride and joy. On the other hand we've done a lot of Broadway shows, again, probably due to the available ambient space here. At the

same time we're doing scoring and all kinds of pop recording."

BUILT-IN QUIRKS

Knowing that Ben has an extensive musical background, one might wonder if all this post work, particularly the voice overs and dialogue, can be somewhat of a rut for a musical person such as Ben. Ben states, "I like to capture music and put it on a film. I don't care what the medium is when there is an intent behind the

process. Dialogue recorded properly is music."

Ben gives us a glimpse of what is in store for the near future at MSA. "Soon, we will be building a two hundred and fifty seat film mixing theater. An interesting aspect about the design of that theater is that it's a departure from my present philosophies about studio design. This theater will have built-in quirks and problems (acoustically) that are in-

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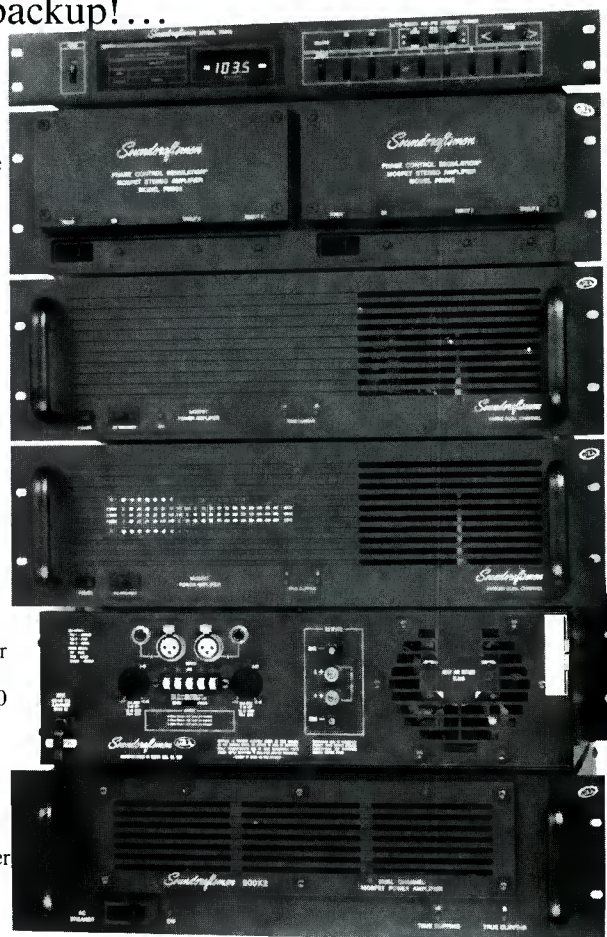
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herent to a normal, stereotypical, conventional movie theater. When mixers work in that theater they will know that they have gotten a mix that "makes sense" in its relationships to the outside world and other theaters. We have already committed ourselves to having a three-position console with as much automation that is technologically available. In addition to the theater we will be building a number of film editing rooms including Moviolas and whatever is necessary in order to facilitate a well-rounded complex that can handle all aspects of post-production. Another item on our shopping list is a hard disk storage medium (tapeless recorder)."

Wonderful things can happen when a solid artistic/musical background and experience is coupled with a strong foothold in technology, physics and science.

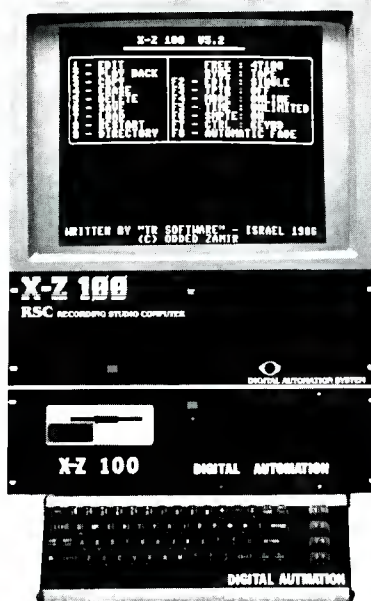
A scientific approach without regard for philosophy is a lost cause. A question most often raised is: Who is this technology for? Well, with all egos aside, the answer is probably whoever wants to take advantage of it. It's nice to know that the recording studio that you are working in can render as much technical information as you need in order to make decisions about the processes that best facilitate the desired goal.

IT'S ALL DOCUMENTED AND AVAILABLE

Ben relates this basic notion to his studio experience, "Every microphone that we have here has been tested with the Techron and catalogued. The point is that there is documentation on every aspect of this studio from control room design to acoustic analysis to equipment specs to you name it, and it's all available to whoever feels that it will be advantageous to their sessions here. We also make ourselves available for guidance and advice if that's needed. David maintains, "while maintaining our audio excellence, we have been very busy developing and maintaining expertise in interfacing different formats, plus various methods and approaches to the work process." Ben concludes, "When all is said and done, what I am really proud of is that we have, slowly but surely, expanded our services and at the same time have not neglected our music clients at all."

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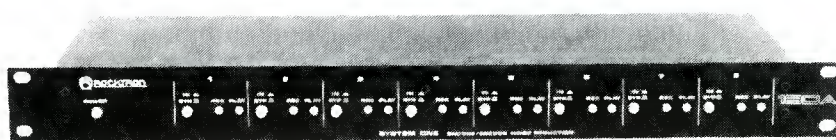
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Sound Reinforcement in South and Central America

IN 1928, JOE FALCON BUNDLED HIS WIFE AND THEIR musical instruments into a Model-T Ford and drove from Louisiana to New York, a journey that took several weeks. When they arrived in the Big Apple, they recorded "Allons a Lafayette," with Joe singing and playing accordion, his wife accompanying on guitar. This occasion marked the first recording of Cajun music, a style born of French people who had left Europe to settle in Nova Scotia. They were eventually kicked out by the Spanish, emigrating to Louisiana, where they settled in Arcadia and adopted the name "Cajun." While their music retained its French character, it also absorbed much from the rich musical heritage of Louisiana. This process of assimilation continues today, as a wide variety of Cajun, Creole and Zydeco groups are beginning to make their mark on the national music scene. Wayne Toups is acknowledged to be one of the top performers of the genre today; his band blends Zydeco and Cajun music with influences of rock, reggae, blues and R&B. The result is music true to tradition yet presented in a modern manner, a perfect example of America's musical melting pot. As a living exponent of the American musical process, Wayne and the Zydecajun Band were selected by USIA (United States Information Agency) to embark on a cultural exchange tour of South and Central America. I was only too happy to accept the job as production manager/sound technician for this tour, which would visit Bolivia, Chile, Paraguay, Brazil, Nicaragua, Honduras, El Salvador and Guatemala during March and April, 1987.

EQUIPMENT DECISIONS

As Wayne's group was heavily amplified, I was most concerned about procuring adequate sound systems in each country. The pattern for USIA tours in the past few years was to carry along as little as possible to save on transportation and rental expense. While the tour itinerary sounded exotic, events such as Rock in Rio, and the emergence of South American pop bands on the international scene, indicated to me that finding reasonable equipment locally might not be that difficult, even for an electric band. Sandra Murphy, USIA programmer for the region, confirmed this hypothesis, so we arranged to procure house PA systems in each country. I drafted a brief tech rider that would be forwarded to each USIA office, describing our sound needs from a frequency response and watts-per-audience-member basis. Wayne owned and operated his own sound system, so he had an excellent grasp of practical and technical sound requirements. Both Wayne and I concurred in the need for consistency in stage sound, so it was agreed that we would carry a monitor system and all stage equipment, including microphones, stands etc. We elected

to go with his monitor setup, as he was already comfortable with it. Wayne's monitor power was provided by a Peavey R-600B mixer/amplifier, and I suggested that we might consider creating a discrete monitor system through the use of mic splits. Wayne could then control the band's monitor mix from on-stage, an idea he took to with enthusiasm! I also included a small 16-channel mixing console, front-of-house electronics, and the various snakes, patch cables and adapters I'd need to tie my equipment into the PA-du-jour. My trusty 28-amp Variac transformer would provide electrical conversion; while we would see some 110-volt power, most places used 220-volt 50 cycle AC. My transformer's large capacity could supply AC for my front-of-house gear, the monitor system, and the band gear handily, with the advantage of maintaining a single power source for all equipment.

Politics can inevitably complicate a tour to foreign countries, and proved to be the case this time. The thought of going to Nicaragua, with the possible consequences real or imagined, proved to be a problem for Wayne: several of his band members balked at going due to the "dangers." I had survived Syria in 1984 and Sudan in 1985, situations I felt were far more serious, with no problems due to expert USIA care in each country. I felt very confident that the trip would be a safe one, but my arguments were not enough to change anyone's mind, so Wayne was forced to make a change in personnel and instrumentation, eliminating his usual keyboard and bass players. For this tour, Wayne Toups and the Zydecajun Band would consist of Wayne Toups, accordion and vocal; Waylon Thibodeaux, fiddle and vocal; Wade Richard, guitar; Randy Ledet, bass; and Troy Gaspard, drums.

MEETING WITH THE USIA

My sound equipment was shipped air cargo from Detroit to La Paz, Bolivia on February 17; I wanted to allow a good two weeks to cover myself against transportation mishaps. I left Detroit on Thursday, February 26, arriving in Washington, D.C. that evening. The following day was spent at the USIA building in conference with Sandra and Leeland Cross, who would function as our USIA escort officer. Lee, a retired USIA employee, was an experienced escort who had spent much of his USIA career in Latin America. Fluent in Spanish, Portuguese, French and Russian, he also knew some of the venues we were to play, advance information I found invaluable. In deference to my previous USIA experiences, Lee didn't bother giving me his standard briefing on the countries we'd visit. We spent most of our time discussing logistics and how we might avoid any travel problems with respect to the equipment. A confident, professional attitude emerged from our meeting; we both knew where we stood, and where our responsibilities separated or overlapped, something that's essential for effective teamwork. Lee would fly to New Orleans on Saturday, February 28, traveling to Crowley, Louisiana to meet

Ed Learned is our resident peripatetic sound man.

Wayne and the group. He planned to brief the band and answer questions about the trip on Sunday. On Monday, he and the band would fly to Miami, connecting with our Eastern flight to La Paz; I was to meet them in Miami.

I spent the rest of my weekend relaxing and visiting old friends and business contacts. I'd engineered for RCI, a top DC-area sound company, a few years ago, working with acts like Whitney Houston, Miami Sound Machine, Lisa Lisa, and handling local festivals and Kennedy Center dates, so I have a lot of people to see in two days. It was with a sense of relief that I flew out Monday morning for Miami; with the travel day, and a rest day after, I could recuperate! Our flight to La Paz was scheduled to leave at 1:40pm, but as South American flights inevitably are, this one was delayed an hour.

TRAVEL WOES

Fortunate, for there was no sign of Wayne, Lee or the band. I phoned Sandra in DC to see if she had any news, but at 2pm they finally showed, with the news that their day had been far from painless. Their flight had been delayed several times, and Eastern had refused to fly several equipment cases despite the payment of excess baggage charges. Lee suspected a case of laziness rather than regulation; we both had seen larger, heavier pieces routinely handled by airlines. We now had no trap case, a smaller drum set and fewer mic stands. After yet another delay due to faulty weather radar on the aircraft, we started our 8-hour flight to La Paz, including a brief stopover in Cali, Columbia. Lee warned us to be prepared for altitude adjustment when we arrived, and I discovered how right he was when I stepped off the plane; I felt very light-headed and short of breath. La Paz is located at 12,000 feet, with the airport, El Alto, at just over 14,000, so you *do* notice it! USIS CAO Tom Carmichael and his staff, who had been waiting patiently to greet us, had us sit and rest while our baggage was collected and customs clearance procured. I found out that my sound equipment had arrived only yesterday; the late arrival was due to a scheduling foul-up with Lloyd Aereo Boliviano. Eastern Air Cargo had delivered the stuff to LAB in Miami, but they had neglected to assign it to a La Paz flight until someone at the US Embassy decided to check up on the overdue shipment! The wisdom of extra shipping time for South American destinations was confirmed by this incident.

Tuesday was a rest day, sorely needed as we still suffered the effects of altitude adjustment. I tried to rest as much as possible, but also spent a lot of time with Wayne, listening to tapes of the group and discussing setup of the band and his philosophy on the music. Wayne had never considered the use of echo or other effects for his live show, so I presented some possible options for him to consider on certain songs. Tom joined us late in the afternoon, and together we contacted Javier Saldias, supplier of the sound system we'd use for our two concerts here. Javier spoke English fluently; it turned out he'd lived in Denver a few years back. We arranged for his gear to be delivered to the theater around the same time the band would arrive. I planned on getting there an hour earlier to set up my house gear, check power, etc.

THE TOUR BEGINS

Wednesday, March 4, the start of our performing schedule, was also the first day I felt close to normal since I'd arrived in Bolivia. I was picked up just after 11:30 by Lee

Cross and a USIA driver, and we had an interesting drive through the narrow, crowded back streets of La Paz, negotiating several hills that seemed to have a 45-degree grade! Our venue, the Teatro Municipal, wasn't that large, seating around 700. It had five tiers of wrap-around balconies in the European opera-house tradition. After confirming that all our gear was present, I checked out AC power. I discovered that the 220-volt receptacles were comprised of 2 110-volt lines. I tied my transformer input tail directly to a power panel off-stage right, using a single 110-volt line. The neutral carried a good 15 volts, so I tied both my neutral and ground tails to a ground lug on the panel. Mandatory equipment for foreign tours is a VIZ Power Line Monitor, a peak-reading AC line voltage meter, which I applied to my transformer output. Input voltage measured a real 109 volts, so I stepped the transformer output up to 120 to ensure adequate level. Despite what I'd heard from Tom, the voltage here was remarkably stable; the house electrician informed me that the theater had its own transformer and voltage regulators, a rarity for Bolivia. I assembled my mix point equipment, finishing around the time the band and Javier arrived to set up (see *Figure 1*).

Wayne Toups played two different diatonic accordions, made by Shine Mouton. Both incorporated high-impedance Shure microphones, permanently mounted on the instrument via brackets. The C accordion used an SM-58 element, while the D accordion used an SM-57 element. The diatonic accordion is a difficult instrument to master, as the number of notes available on each is limited with respect to key. The type of accordion Wayne used was dictated by which key the song was played in; on tunes where the key was modulated, he would have to switch instruments during the song. The outputs from these accordions were run into Audio-Technica direct boxes, then into my snake system. Wayne depended on the monitor system to hear himself, as he didn't have an on-stage amplifier like everyone else. He carried two Shure SM-58 low-impedance mics, for use on vocals. I convinced Wayne to try one of my E-V ND-757 mics for vocal, and he fell in love with it immediately! In one sense, the E-V was a little *too* good: Wayne's vocal position was directly in front of the drums, so the increased sensitivity of the 757 resulted in greater drum leakage. This posed a problem, since the Municipal Theater was a bit on the live side, with a reverb time of close to 2 seconds. We decided that we'd use the E-V on his voice at outdoor gigs, or venues where the acoustics were less lively. Otherwise, we'd stick with the SM-58.

Fiddle player Waylon Thibodeaux was the group's youngest member, and a former Louisiana junior fiddle champion. He carried two Italian fiddles, equipped with Barcus-Berry pickups. The pickup was run into a Yamaha SPX-90 digital effects processor, then to an Aerial direct box, with the signal fed to his Peavey Nashville 400 amplifier. Waylon used the SPX-90 in the pitch change mode only, operating it via a foot switch. This let him change the pitch of his fiddle for certain songs without having to retune or switch to his backup fiddle, which neither he or I thought sounded all that great. Waylon also used an SM-58 for his vocal, and preferred it to a 757 after comparing the two.

Guitarist Wade Richard was a veteran of many Louisiana bands, and contributed rich blues and R&B shadings to the Zydecajun band. He played an Ibanez hollow-body guitar, which he ran into an Ibanez multi-effects box. This was a rack-mountable unit, which he located on top of his Peavey

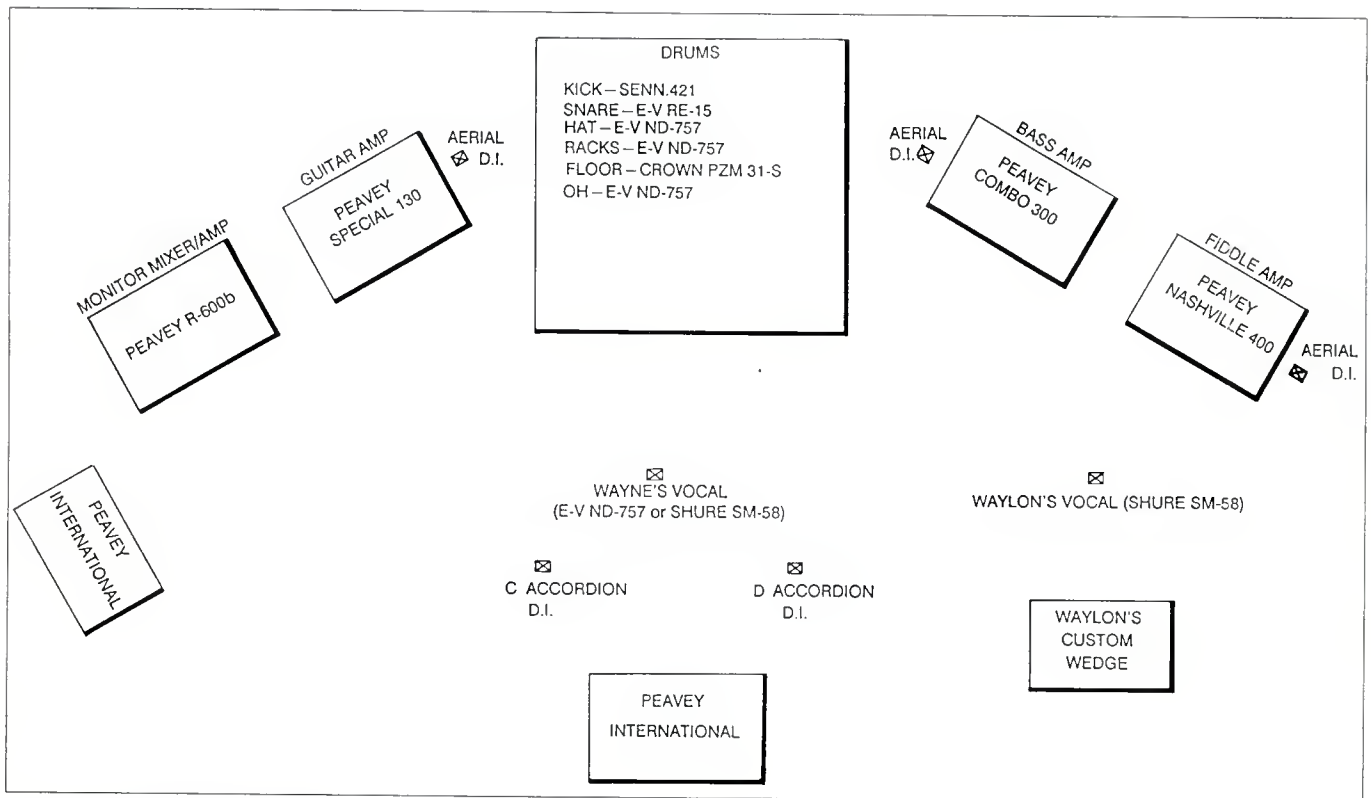


Figure 1. The setup for Bolivia.

Special 130 amplifier and operated via a remote foot switch. He could use effects like delay, chorus and compression singly or in conjunction. I used an Aerial direct box on his amp, using the amplifier's extension speaker output into the D.I.'s padded input. This gave a rich, warm guitar sound without the use of a mic.

Proper presentation of the rhythm section was one of Wayne's major concerns. I had to have independence in level and tonal control of the bass, so I used an Aerial direct box straight off the instrument. Randy Ledet played a Peavey TKO bass, amplified with a Peavey Combo 300 amplifier. I had him run his instrument levels full out, and make any volume or tonal adjustments dictated by room acoustics on the amplifier; my signal would be consistent from gig to gig. Troy Gaspard played a Tama drum set, with two mounted toms and a single floor tom. I used a Sennheiser 421 on the kick, placed off-center with the mic's

head just slightly inside the hole Troy had cut in the front head (Figure 2). The snare was mic'd from the top with an E-V RE-15. I used an E-V ND-757 on the rack toms, placed between the two drums, and placed a Crown PZM 31-S on the floor directly underneath the outside rim of the floor tom. The floor tom sounded very rich, yet the PZM picked up enough of the cymbals, so an overhead wasn't required for a smaller place. I used an E-V ND-757 on the hi-hat, where I found the excellent high frequency response of this mic to be a real advantage. For less reverberant venues or outdoor gigs, I would add another 757 as a real drum overhead, placed over the rack toms but cheated toward the ride cymbal side of the set.

The band's on-stage monitor system was comprised of Wayne's two Peavey International cabinets and a customized wedge that belonged to Waylon. One International was placed directly in front of Wayne, placed far enough away from his vocal position so he had 3-4 feet of coverage on either side of him. Not only did this allow him to move around a bit and still hear, but Randy could also get some coverage. The other Peavey was placed just downstage of Wade's amp and angled in toward Troy. This "side fill" configuration gave us decent coverage for Wade, Troy and sometimes even Randy if he wandered slightly stage right. Waylon used his wedge as his own monitor. The R-600B has 6 channels, so Wayne and Waylon's voices, the two accordions, fiddle and guitar comprised the monitor mix. Vocals and accordions were on top in the mix; fiddle and guitar were mixed in direct proportion to how much I had them turn down their individual amps. We could keep the amps, pointing at the audience, turned down to keep audience stage volume low, but the guys could still hear on stage.

My house PA system for the La Paz concerts was a combination of house components with a Javier augment. The



Figure 2. The drum-set mic'ing.

theater had a pair of Yamaha S4115H cabinets; Javier brought two Peavey SP-2 cabinets and a Randall power amp. I used one of each type cabinet per side, with the Peavey on top, horn up. Each side's stack was y-d into one side of the Randall, which purportedly delivered 400 watts/channel into 4 ohms. By running my house gear at line level, I was able to drive this conglomeration with no problem. The band performed around seven songs, so I could get my feet wet and Wayne could squeeze in a band rehearsal. I came up with some tasteful echo and reverb things with the SPX-90 that Wayne really liked, so I saved those programs for later use. Javier and his two assistants had never seen the unit before; I ended up giving him a half-hour seminar after sound check! We were also invaded by a local TV crew, who asked for, and got, several more songs.

THE FIRST SHOW

I think everyone was a little apprehensive about the first show. It was the Zydecajun Band's first ever concert outside of the U.S., first concert with this line-up and the first performance of a Cajun band in Bolivia. The crowd was small, quiet at first, but warmed to the group quickly. The band charged right through the show with amazing energy; Wayne was all over the stage, spinning around and dancing while he played, and Troy took charge on drums. The altitude still posed a problem; both Wayne and Troy made frequent use of a small oxygen tank Tom had placed just behind the drums. By the end of the show, the "quiet, reserved Bolivian audience" was anything but, applauding loudly and calling for one more, which an exhausted Wayne was somehow able to deliver with elan.

Thursday featured a double shift at the Municipal Theater; we rehearsed in the afternoon and played another concert that night. Wayne spent a good two hours tightening up arrangements with his still-new band. I wasn't happy with our fiddle sound so far, and wanted to try some EQ and volume changes; Waylon and I worked on it until we had something we both liked. I had a chance to spend more time on stage, helping to tighten up the monitor mix. I also invited Wayne out front during Waylon's "Talk Talk Song," an exercise in vocal gibberish and hot lickin', to have a listen and offer constructive criticism. He liked what he heard, and also passed along compliments from Javier and other local musicians about last night's show. It always feels good to know you're on the same page, but I felt I could do even better... and did during the evening's concert. We were all more comfortable with the sound, and starting to feel normal physically. The band's execution was vastly improved, and the monitor mix was right in the pocket. The crowd had doubled, due in part to positive press coverage praising the band's rhythmic drive and Wayne's unbridled enthusiasm on stage. I was more comfortable with the material now that I'd done a show, and by the concert's finale I felt I had a handle on what I wanted to do mix-wise for the remainder of the tour. The band was inundated by autograph seekers after the show; Tom said he'd never seen it like this. Bolivia liked Zydecajun in a big way!

FINALE IN BOLIVIA

Friday, March 6 featured a morning tour of the Andes Mountains outside of La Paz city conducted by Javier; Randy, Waylon and I took him up on it, and it was spectacular! We caught a late afternoon flight out of La Paz bound for Cochabamba, accompanied by Tom and cultural



Figure 3. The band in Cochabamba in front of an ad for the show. L to R: Troy Gaspard, Waylon Thibodeaux, Randy Ledet, Wade Richard, and Wayne Toups in the wild shorts.

assistant Silvia Vizcarra. This meant flying mountains, some as high as 20,000 feet, and entering the interior of Bolivia. Located in the "foothills," Cochabamba was at lower altitude, looking more "green" than La Paz. The temperature was much warmer, and a full-scale downpour hit us while on the way to the hotel. Once checked in, Silvia drove Lee and I over to the Teatro Acha so we could have a look at it. The theater looked like an old church, with high vaulted ceilings over both the audience and the stage. Addition walls and a proscenium opening created a "stage" where the altar area had once been. There were two wrap-around balconies, and total capacity was around 600. Acoustics were horrid; the rooms reverb time was in excess of 3 seconds, due to wood and plaster surfaces and the high ceiling. The house power panel was off-stage left: a wooden board with switches, single fuse receptacles and wire all tangled together and running everywhere! I'd brought my meter, and discovered that this place had a single 220 hot, a questionable neutral and no ground. I found a bathroom backstage about 20 feet from the panel, and made a mental note to get my ground from a pipe there.

Saturday was our concert day. The morning was spent shopping in the town's main square (Figure 3). I got some 14 gauge stranded wire for my day's electrical project, which began at 2pm. I measured a good 20 volts minimum on the neutral, so I tied both my neutral and ground tails to the bathroom water pipe, using two separate wires for safety in redundancy. My sound system, provided by Mario, a local guitarist, consisted of two E-V 2-way cabinets, two E-V 3-way cabinets and a Peavey Decca 700 power amp. That was plenty of PA, as the only hope for clarity of sound was to keep the volume as low as possible! We had over 2/3 of the house full, which soaked up just enough of the reverb to make the sound passable. The AC voltage fluctuated wildly during the show; I measured drops of 20 volts from time to time. The band was again called back for an encore, and was deluged with autograph seekers after the show.

Sunday started way too early. We had to leave the hotel at 5:30am to make our 7am flight to Santa Cruz, regional capital of eastern Bolivia. The change in climate and topography was remarkable; Santa Cruz was hot and tropically humid, palm trees dotted the landscape. Local assistant Alberto Gamarra had us ensconced in our hotel almost immediately; he informed me that a large Peavey PA had been procured for us. Setup at the auditorium of the Universidad Gabriel Rene Moreno was scheduled for 2pm, so

we all had a chance to grab some needed rest. The hall was small, with a capacity of 400. Soft seats, plush carpeting and acoustical treatment on the ceiling conspired to reduce reverb time to around a second, just perfect for our electrified group. The master power panel was about 50 feet off-stage right, and had a good ground, single 220 hot, and the first clean neutral I'd seen in Bolivia. Two Peavey 115 HD bass bins and four Peavey CH-4L radial horns comprised the sound system, with Peavey CS-800 and CS-400 amps handling lows and highs respectively. The CS-800 incorporated Peavey's plug-in crossover cards for bi-amp use, frequency set at 800 Hz. The seating area was raked upwards very steeply, so I made sure that I panned the top horn on each side upwards to throw real highs to the back. I also backed the level of the CS-400 down so 4 horns would properly match up with two woofers. The concert was fabulous; the place was sold out, and the young student audience got right into it, yelling and applauding wildly. It was a full-scale party, resulting in two encores. I had the house lights turned on during the last song, and when the band saw this energetic crowd dancing in the aisles and going nuts, they played with a special intensity. The guys were mobbed after the concert, and many waited for the band outside the auditorium. The band members were whisked away to various parties by friendly Bolivians; Lee and I, along with our USIS friends and some audience members, frequented one of the many sidewalk bistros on the town square for dinner, wine and post mortems on our Bolivian performances.

ON TO CHILE

Monday found us traveling back to La Paz, where Tom hosted a farewell party for us at his home. On Tuesday, March 10, we left La Paz at 9:30am on a flight to Santiago, Chile. We made a brief stop in Arica, the northernmost city in Chile, where we encountered a different type of landscape: desert, as far as the eye could see, from the horizon to the waters of the Pacific. We arrived in Santiago, Chile's capital city, around 1pm, and were greeted by cultural assistant Daniela Muller. Our baggage was quickly assembled, but clearing customs was another matter. It took over an hour, as the customs officials wanted to inspect each piece and check all serial numbers against those on our manifest. We eventually escaped and drove the half hour into Santiago, a very cosmopolitan city with beautiful wide boulevards. Once checked into our hotel, located on the Plaza de la Constitucion near the Presidential Palace, we attended an informal briefing on Chile given by Daniela and Dan Saint-Rossy, who would be our USIS control officers during our stay in Chile.

Wednesday was essentially a day off, save for a noon press conference at the USIS Bi-National Center, located a few blocks from our hotel. USIS-Santiago had suggested that the band play a few numbers, and Wayne agreed. We decided that a full setup in the room provided would be undesirable due to the small space available, so we unpacked the minimum amount of gear. Troy used only his bass drum, snare and high hat, while Wade and Waylon shared one amplifier. I set up one vocal mic, and used the two International cabinets, one as PA, the other as monitor. The room was so lively that we had to cover the amps with packing blankets, and even put a towel over the snare drum to lower its level. By noon, we had only one half of the invited journalists present, and wondered why, until Daniela arrived with the news that the city center had been sealed off

by the army and police! No, it wasn't a coup; Augusto Pinochet, President of Chile, had decided to make a surprise address to the government, so for security reasons the area around the Palace was sealed off. No one could come in or out until the President was secure at his destination. Rather than wait, Wayne performed for the people in attendance, then did another couple of songs an hour later when the rest of the press arrived. Though the group performed without their usual stage presence, everyone was impressed with Wayne's emotion when singing. This projection of feeling made him "sympatico," something Chileans could relate to from their own feelings. Though unfamiliar with Cajun music, most of the journalists felt that the music would enjoy tremendous success here, especially with young audiences. This buoyed everyone's expectations, as three of our four concerts in Chile would be to college audiences.

Voltage was very stable, and a good 50 amps were available here.

A VENUE AT THE MOVIES

Thursday, March 12, marked our first performing foray into Chile. We left Santiago in the early AM, traveling to Concepcion, a city on the Pacific coast about 300 miles south of Santiago. Mercedes Pujol, director of the local Bi-National center, met us at the airport and delivered us to our hotel, where we sorted out the day's scheduling problems. Our venue, the Teatro de la Universidad de Concepcion, was an operating movie house, so we would have to work our setup and sound check around the movie screening schedule. We couldn't set and leave the band gear, as it would block the screen. With two hours to go before the first screening, we hurried over to the theater to assemble our gear, pulling it behind the screen for the moment; we'd return that evening to do the final placement and sound check. I found a power panel near the dressing rooms, upstairs stage left, to tie in my transformer tails. Power was 220 volts on a single hot, with a clean neutral and good equipment ground. Voltage was very stable, and a good 50 amps were available here. The theater seated approximately 1000, and had a single balcony facing the stage. The acoustics were extremely dead; reverb time was only half a second, due to curtains hung from every wall and thick carpeting. Sono, a local PA company, arrived shortly after we did, and once I had my house gear assembled I supervised placement and tuning of their system. Per side, the system was 2 Yamaha 18-inch woofers in a horn-loaded cabinet, 2 Nobis 15-inch speakers (Brazilian) in horn-loaded cabinets, and 2 Chilean-made radials with Selino (Brazilian) compression drivers. Power amps were a QSC 3050 for bass, Yamaha P-2200 for mids, and either a Yamaha P-2050 or a Tapco CP 500M for highs. The crossover was a Yamaha F-1030, points set at 250 Hz and 1.25 kHz. This system had plenty of power, but didn't match up very well internally. I changed the upper crossover point to 1 k, as it was obvious that the 15-inch speakers couldn't cut it that high up. I also backed the volume of the mids and highs off (the crossover had been fully up everywhere), and used my graphic to smooth out the mids and highs. While this was going on, Wayne and the group were conducting an im-

promptu TV interview for a crew that appeared out of nowhere; the clip would be on the evening news before our concert, perfect publicity for us. As soon as I finished tuning the PA, the TV guys descended on me for a short interview, with Lee acting as my interpreter. I was quite surprised by the whole thing; there went my anonymity! We returned at 8pm for sound check, only 45 minutes before the house would open. We hurriedly set up, and completed our sound check in only 15 minutes. The hall was so dead that I had everything full up in the mix for the first time; I took advantage and taped the performance. We had a sell-out, with the young student crowd wasting little time in rocking the place. Inspired by the exuberant reaction, the band played the best show to date. I'd pushed the SPX return more at this show due to the dead quality of the room, so the effects were more audible as we listened to the tape at our after-show party. Wayne was so knocked out by them that he asked me to come up with some other ideas to spice things up even more. Troy and I discussed some possible gated reverb and reverse room effects that we'd incorporate on the drums in response to Wayne's request.

The PA in Concepcion had been more than adequate for our needs; little did I know that I was about to swing to the other end of the scale! We returned to Santiago late Friday morning, and I went with Dan over to the University of Chile to supervise the early afternoon setup. Our gear had been delivered, and I noticed 2 Yamaha 3-way studio monitors and a Sony hi-fi receiver on one side of the stage—no doubt left over from a morning lecture. How wrong I was; the house electrician informed me that *was* the PA. Dan was shocked, as the University had assured him adequate sound would be provided. It was now too late to arrange anything else, so we had to improvise. We used our 2 International cabinets as PA, powered by the amplifier of our R-600B monitor controller. The 2 studio monitors and Waylon's wedge comprised the monitors, with the mixer section of the R-600B feeding the Sony receiver, which would function as monitor amp today. The Sala Isidora Zegers in the Faculty of Arts was a small hall, seating only 400, with a single balcony. Acoustics were good, due to curtains covering the rear wall and acoustical treatment on the side walls and ceiling; reverb time was just over a second. I let the drums go acoustic, amplified vocals and accordions, and mixed in the rest only on solos. Monitors were another matter. At the level Wayne needed, the monitors distorted; the Sony couldn't cut it. The only solution was to play as softly as possible. We had another sellout, with U.S. Ambassador Harry Barnes among the dignitaries in attendance. Several audience members called out requests to Wayne in French; he got a big kick out of this. The group did an excellent job under trying circumstances, though both Wayne and Waylon were quite hoarse after the show: a symptom of singing louder to compensate for the weak monitor system.

VOYAGE TO VALPARAISO

Saturday began with a 3-hour drive to Valparaiso, a major Chilean port city. We drove past some of Chile's finest vineyards on the way; Lee, Dan and Pedro, our driver, agreed they produced the finest wine in South America. Our first stop was the Universidad Federico Santa Maria, where we were to play the Aula Magna. It seated 1300, and acoustics here were of the barn variety: a reverb time of 2 1/2 seconds and very bright due to a plethora of plaster surfaces. The PA was 2 Yamaha

powered columns and a floor monitor per side, a system barely adequate for the room. Dan accompanied the band to our hotel, located in the neighboring beach resort of Vina Del Mar, while Lee and I waited for the house electrician. When he finally showed, it took almost 2 hours to get correct power and the proper wiring. When I asked for a hot and neutral, I got 2 110 volt hots. When I asked for a single 220 hot, I got a 380 volt hot. Everywhere I checked, the neutral carried at least 30 volts. A single 220 hot was finally produced from a back-up power board high on a lighting access walkway off-stage left, and I tied both my neutral and ground to a neighboring water pipe to ensure 0 voltage. I had to boost bass on my graphic to get any kind of lows from the PA, and found that by pulling down 2 k, I could reduce the brightness problem. We had only 1/2 a house for our concert that night, and the reaction was not as impassioned as we'd seen at our other Chilean concerts.

Sunday was a free day, spent on the gorgeous beaches on Vina Del Mar, admiring the local scenery and swimming in the fantastic surf. The water was *very cold*, due to the Humbolt current which brings water from Antarctica up the western coast of South America; I'm accustomed to Michigan lakes so I could stand it. Enjoying this gorgeous day, we could understand the sparse turnout at last night's show—they'd rather be at the beach! We returned to Santiago around sundown, when Lee, Dan and I called our local promoter to confirm adequate PA; we didn't want a repeat problem. Monday's concert was held at the Teatro Oriente, another operating movie house. The screen here was elevated 5 feet, so we could set up, sound check and leave everything without blocking sight lines. Acoustics were perfect for us: a fairly large room, seating 1100, but a combination of drapes and carpeting cut the reverb time to just under a second. Per side, the PA consisted of an 18-inch, a 15-inch and a Peavey radial horn. The cabinet design for the 18-inch and 15-inch speakers was identical to the Concepcion system. Two Peavey CS-800 amps, equipped with Peavey plug-in crossover cards, and one Peavey CS-400 power amp supplied PA power. A 24-channel snake had even been provided, so I didn't have to use mine. AC power came from a panel just off-stage left, yielding a single 220-volt hot and a clean neutral and equipment ground. We had a longer-than-normal sound check while Wayne worked in some new songs, but still finished easily by 1pm, when the theater had to be cleared for movies. The evening's concert provided a great end to our Chilean concerts. The crowd of 800 was comprised mostly of students; in the balcony, they were dancing from the first song and never let up! Wayne pulled a couple of girls on stage to dance with him, which really got a reaction! I had a chance to try out the new drum effects, which worked well; the local PA guys came running over to see how I was doing it! The audience was all over the band for autographs after the show, and the party continued back at the hotel where the group and various guests enjoyed the tape I'd made of the evening's show.

LAST STOP PARAGUAY

Tuesday, March 17, found us travelling to a new country: Paraguay. We cleared customs in Santiago easily this time, and enjoyed a comfortable 3 1/2 hour flight to Asuncion. We were met by Public Affairs Officer Alan Rodgers and Jeff Brown, USIS control officer for our visit. The airport was modern and air-conditioned, something we appreciated as it was a good 90 degrees outside! Customs clearance was expeditious, and Alan escorted Lee and the group



Figure 4. The sound system at the Coca-Cola plant in Asuncion, Paraguay.

to the hotel. Jeff and I took a little detour at my request: I wanted to see the site of the outdoor concert we'd play here on Thursday. It was on the grounds of the local Coca-Cola bottling plant, located directly behind the plant. The stage was set against the back wall of the plant, facing across 50 feet of tarmac to another wall. This area was completely covered by a tin roof, with the remaining two sides open. My acoustical worry was not reverb but an early-reflection problem: when I clapped my hands I got a series of closely-spaced discrete echos. Power was available from a small panel on the plant's back wall; I was told this could provide 50 amps with a single 220 volt hot, neutral and functional ground. We adjourned to the hotel, where I proceeded to enjoy the rooftop swimming pool and a pretty view of Asuncion, lush and green, situated beside the River Paraguay.

Wednesday's schedule included a concert at the USIS Bi-National Center Auditorium, a very small room seating about 300. The auditorium had slightly sloped seating, a carpeted floor and some acoustic treatment on the walls, with a reverb time of 1 1/2 seconds. Power was supplied from a panel off stage left, but I quickly discovered it had no ground and a neutral that carried 40 volts. I found the building's master water pipe only 25 feet away; after scraping off paint to expose bare metal I tied both my neutral and ground tails here, and used the single 220 hot available at the power panel. PA was a pair of Peavey SP-1 cabinets and a Peavey CS-800 power amp, plenty for the hall. I decided not to mic the drums here, a decision that proved wise once we started sound check. The room was so small that I had the group play as soft as they could on stage, mixed the vocals and accordions up in the house, and added the other instruments only on solos. I did have one exciting moment during the show however — Wayne's vocal mic cut out in the middle of the first song. I could hear vocal in the monitors (the room was *that* small), so I knew the mic and cord were okay. I patched in a different snake channel while the band played their solos, ran up on stage to make the change there, and returned to the console in time to catch Wayne when he stepped back to the mic for the last verse. Our sold-out crowd contained all age groups, and was very appreciative. The students in attendance told us that they had expected a "folklore" concert, with a lot of twangy guitars and old-time songs, not a rocked out, exciting show. They promised to spread the word for tomorrow's show. My earlier problem turned out to be a broken wire in the pigtail end of the snake, which had been touch-

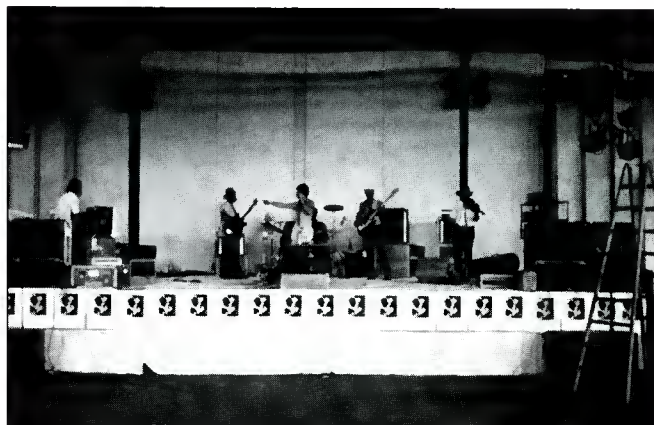



Figure 5. The band doing soundchecks for the show in Asuncion.

ing just enough to make it through sound check, yet fail for the show. It was easily repaired with a quick soldering job. I talked with the sound technician about gear for tomorrow: we arranged to more than double the amount of PA.

FINISHING WITH A FLOURISH

Thursday was our outdoor concert on the Coca-Cola grounds. We began our setup around 2pm, finished around 5. Per side, I had 2 Peavey SP-1 cabinets, a Peavey 3-way component stack and an Ampeg bass cabinet (Photo 3) as PA. It was a real assortment all right, and getting all of it working was proving to be a headache for the local technicians. I'd asked for an additional Peavey CS-400 for use as a monitor amp. I could now use the monitor output on the R-600B to power another amp, creating another separate monitor mix. Main mix was assigned to Wayne, monitor mix to Waylon and the "side fill." The Coke electrician ran a power board over to the stage, tapped off the panel I'd seen Tuesday. I got a single 220 volt hot, but discovered that the ground was no good and the neutral carried over 10 volts. I found a nice metal pipe on the back wall about 50 feet from the stage, and tied both my neutral and ground to it via separate wires; Jeff had alertly thrown the spare wire in his car. That cleaned me right up, and cleaned up the local PA when I suggested they tie the chassis of their amp rack to my equipment ground. Our hum problem vanished, and buzzes dropped to a point where I could begin sound check. I went with my full mic'ing, and having the extra monitor mix really tightened up the stage sound. We had a crowd of around 900 that night, mostly young people, and it didn't take long to become a full-scale dance party! Wayne had his blues, rock & roll and R&B covers worked into the set with great effect, and played the longest show of the tour. Wayne's original, "Zydeco Shoes," closed the show with a mad, dancing climax. After two encores, the band spent another 45 minutes autographing posters and programs for the adoring crowd, who didn't want to let the band go! PAO Alan Rodgers said the concert probably did more for U.S.-Paraguay goodwill than a year's worth of regular programming.

NEXT ISSUE: The tour continues, visiting Brazil (Sao Paulo, Campinas, Brasilia, Salvador, Rio de Janeiro), Nicaragua (Managua, Granada), Honduras (San Pedro Sula, LaCeiba, Tegucigalpa), El Salvador (San Salvador) and Guatemala (Quetzaltenango, Guatemala City). 

Notes on 70-volt and Distributed System Presentation

DREW DANIELS

The so-called 70 volt line distributed loudspeaker system wiring scheme offers a flexible means of operating multiple loudspeakers connected to singular amplifier lines.

The definition of the "70 volt" system is one in which 70 volts (70.7 volts) represents the maximum operating VOLTAGE delivered from the driving amplifier, regardless of the particular power level capability of that amplifier. A "70 volt" speaker transformer with power level taps of 1, 2, and 4 watts, will draw 1, 2, or 4 watts, depending on the tap selection, when the line voltage fed to the transformer's primary reaches 70.7 volts.

An amplifier capable of developing 70 volts into a load of 8 ohms can be used to provide 600 watts in a 70 volt system

The 70 volt and other constant voltage (e.g. 25 volt, 50 volt, 140 volt) systems were devised to provide an economical means of driving many speakers over long signal lines with low loss. Higher voltage on the line allows use of less current in the wire, which in turn causes less voltage drop and power loss in the wire itself and allows use of smaller less expensive wiring.

It is not necessary to achieve 70 volts in the speaker lines to successfully operate a 70 volt system, but following the same logic that applies to any amp and speaker combination, the square of the voltage divided by the number of ohms representing the total system load will determine how much power will actually be distributed through the system.

An analogy of distributed system operation can be made from everyday house wiring to illustrate how a distributed system works: in a house there is an electrical conduit carrying 120 volts all over the house to wall outlets. A 20-amp circuit breaker feeds the line. At any outlet you can plug in a lamp to give you as much light as you need in that particular location, however, since the line is supplied by a 20 amp breaker, you can only plug in 2400 watts (120 volts X 20 amps) of total load before you run out of power and trip the circuit breaker. You can use twenty-four 100-watt lamps or forty-eight 50-watt lamps or a hundred 24-watt lamps and so on, to use all of the available power, but you

might also only use one lamp in each room drawing only a few hundred total watts, which will leave power to spare. The distributed system is a constant voltage system.

Loudspeaker sensitivity and impedance rating play a big part in overall system efficiency.

An amplifier capable of developing 70 volts into a load of 8 ohms can be used to provide 600 watts in a 70 volt system. This much power might be used to drive 200 ceiling speakers each with their transformer taps set to 3 watts, or half of all the speakers set to 4 watts and half set to 2 watts to create a loud zone-quiet zone arrangement where the two zones differ in sound level by 3 dB (3 dB is half/twice power and a just noticeable difference in speech sound level).

Substituting an amplifier with a maximum 50-volt / 4-ohm (600 watts) load capability rating and doing nothing else, would drop the available power to this system to 300 watts and provide each speaker in the system with half the power indicated by its transformer tap setting. Since this substitute amplifier is rated to drive a 4-ohm load where the original amplifier was rated at 8 ohms, another 200 speakers—a doubling of the original number—could be added to the system and would be driven at the same power level as the original 200 units, or half the rated tap setting value, allowing the full 600 watt potential of the substitute amplifier to be realized.

The Ohm's Law-based equations provide an easy way to determine just how much voltage, current or power is involved in particular system designs or what the total loading on a distributed line will be based on the wattage taps used and number of speakers connected to the line. JBL tech note, Volume 1, Number 2: "70-volt Distribution Systems Using JBL Industrial Series Loudspeakers," gives tables and other valuable information to aid in distributed system design.

Loudspeaker sensitivity and impedance rating play a big part in overall system efficiency. Speakers of different impedances draw different amounts of power from a constant voltage (e.g. the 70-volt system) source. For example, let's use two commercially available speakers, A and B. The pertinent specifications of the two devices are as follows:

SPEAKER A: Sensitivity = 97 dB SPL, 1 W, 1 m and impedance = 8 ohms.

SPEAKER B: Sensitivity = 86.5 dB SPL, 1W, 1m and impedance = 6 ohms.

Drew Daniels delivered this article as a paper before the NSCA, September 10, 1985.

Speaker transformers have insertion loss that is due mostly to resistive losses in the transformer, so the transformer loss itself can be calculated as if the loss element is a resistor. If we know that some typical transformer has one dB of insertion loss when working into its rated load impedance (usually 8 ohms), then we can calculate backwards and find the transformer's equivalent resistance to be 2 ohms. We know this from the fact that a transformer that has 1 dB of loss delivers 4 watts to a speaker when its 5-watt tap is used.

The difference between the 4.69 watts for speaker B and the 4 watts for speaker A is only a little over one-half dB.

A speaker with lower impedance will draw more power from a constant voltage source, and if the source had negligible resistance itself, then the 4 watts available to the 8-ohm speaker A would become 5.3 watts but it's not quite that simple. If we place speaker A in a series circuit with our typical transformer, we find that the speaker drops 4 watts and the transformer drops 1 watt to make up the 5-watt total. The current across this combination is 0.707

ampere, which means the voltage drop across the 10-ohm load (8 ohms for the speaker and 2 ohms for the transformer's resistive loss) is 7.07 volts.

Substituting speaker B across the same constant voltage produces 0.884 ampere of current through the load (now 8 ohms total), and causes 1.56 watts to be lost in the transformer and 4.69 watts to be delivered to the speaker.

The difference between the 4.69 watts for speaker B and the 4 watts for speaker A is only a little over one-half dB. What might have seemed to be a potential advantage is eaten up by the transformer, and worse, the lower impedance speaker B is now pulling 6.25 watts from the line, which is 25 percent more power and will mean that you can only connect 80 percent of the number of speaker A's you would have been able to connect to the line before you exceed the amplifier's available power.

The issue of speaker sensitivity is much more important when many speakers are used and the "dB's for dollars" problem can eat up profits quickly. Speaker A offers 97 dB SPL for 1 watt at the standard 1 meter distance. Speaker B offers 86.5 dB SPL for the same watt. This means that to achieve the same sound level at the same distance, speaker B will require more than 10 times more power than speaker A. Another way to look at it might be that it will require at least 3 times as many of speaker B to provide the same sound level as speaker A, and in some physical situations up to 10 times as many of speaker B.

OHM'S LAW-DERIVED EQUATIONS

TO FIND WATTS:
(volts squared) divided by ohms
(amps squared) X ohms
volts X amps

TO FIND AMPS:
volts divided by ohms
watts divided by volts
square root of (watts
divided by ohms)

TO FIND OHMS:
volts divided by amps
(volts squared) divided by watts
watts divided by (amps squared)

TO FIND VOLTS:
amps X ohms
watts divided by amps
square root of (watts X
ohms)

A Basic Program to Figure 70 volt Systems

For MSDOS Computers

```

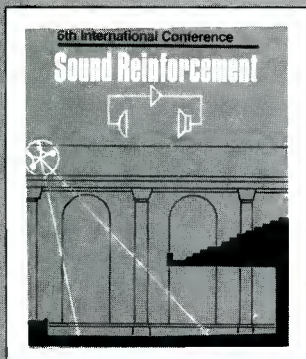
1 CLS : COLOR 15: PRINT : PRINT
2 PRINT "          Drew's 70-volt System Calculator Program": GOTO 4
3 CLS
4 PRINT : PRINT
5 PRINT "  What is the speaker line voltage (the amplifier's actual output voltage)?"
6 PRINT : INPUT "          "; V: IF V = 0 THEN 1: PRINT
7 PRINT "  What is the amp's power rating (or the power you want to supply to the line)?"
8 PRINT : INPUT "          "; W: IF W = 0 THEN 8
9 R = (V ^ 2) / W
10 IF R < 4 THEN 44 ELSE 11
11 CLS : PRINT : PRINT : PRINT : PRINT : PRINT
12 PRINT USING "  For ###.# volts at #### watts, the line impedance is ###.# ohms"; V, W, R
13 COLOR 14
14 PRINT "          and will drive the following number of 70-volt speakers:"

```

```

15 PRINT : COLOR 15
16 A = 20000 / R: B = A / 2: C = A / 4: D = A / 8: E = A / 16: F = A / 20: G = A / 32: H = A / 40: I = A / 60: J =
A / 64
17 IF R < 2 THEN 44 ELSE 18
18 PRINT " Tap Tap Tap Tap Tap Tap Tap Tap Tap Tap "
19 PRINT " 0.25 W 0.5 W 1 W 2 W 4 W 5 W 8 W 10 W 15 W 16 W"
20 PRINT " -----"
21 COLOR 14
22 PRINT USING " ##### "; A,
B, C, D, E, F, G, H, I, J
23 COLOR 11: PRINT "Power in watts absorbed by each speaker:"
24 E = V ^ 2: COLOR 10
25 AP = E / 20000: BP = AP * 2: CP = AP * 4: DP = AP * 8: EP = AP * 16: FP = AP * 20: GP = AP * 32: HP =
AP * 40: IP = AP * 60: JP = AP * 64
26 PRINT USING " #####.## "; AP, BP, CP, DP, EP, FP, GP, HP, IP, JP
27 COLOR 15
28 PRINT : PRINT " Press R to run again, press I for info, press Q to quit."
29 E$ = INKEY$: IF LEN(E$) = 0 THEN 29
30 IF E$ = "Q" OR E$ = "q" THEN SYSTEM
31 IF E$ = "I" OR E$ = "i" THEN 33
32 IF E$ = "R" OR E$ = "r" THEN 3 ELSE 29
33 PRINT : PRINT : COLOR 15
34 PRINT "The voltage you enter might need to be trimmed to account for line loss or"
35 PRINT "insertion loss in transformers. Remember that when small transformers"
36 PRINT "are driven past their nominal line voltage rating, they exhibit non-linearities"
37 PRINT "due to transformer core saturation, and will draw more than the indicated power"
38 PRINT "and convert less of it to sound. The system then becomes unreliable and pro-"
39 PRINT "duces undesirable load conditions for the driving amplifier."
40 PRINT : PRINT : PRINT
41 PRINT " Press any key to continue."
42 X$ = INKEY$: IF LEN(X$) = 0 THEN 59
43 IF X$ < > "" THEN GOTO 3
44 CLS : PRINT : PRINT : PRINT
45 PRINT "THE LOAD IS LESS THAN 4 OHMS! CONSIDER ONE OF THE FOLLOWING STEPS:"
46 PRINT
47 PRINT "1 - Reduce the number of speakers on the line."
48 PRINT
49 PRINT "2 - Set some of the speaker transformer taps to lower wattage."
50 PRINT
51 PRINT "3 - Increase the line's nominal maximum operating voltage."
52 PRINT
53 PRINT " NOTE: do not increase operating voltage beyond the speaker trans-"
54 PRINT " former input rating. The system will be unreliable and the"
55 PRINT " speakers will draw more than the indicated power shown on "
56 PRINT " the speaker transformer taps."
57 PRINT : PRINT : PRINT
58 PRINT " Press any key to continue."
59 X$ = INKEY$: IF LEN(X$) = 0 THEN 59
60 IF X$ < > "" THEN GOTO 3

```



AES 6th International Conference

Sound Reinforcement

STOUFFER HOTEL

Nashville, Tennessee, USA

1988 May 5-8

Ted Uzzle, Chairman, Altec Lansing Corporation, Oklahoma City, OK, USA

An intensive 3-day conference dedicated to the technology of sound reinforcement and architectural acoustics

For many years sound reinforcement has been a vital subject of technical sessions at the AES conventions. Now the AES 6th Conference, the latest in a series of conferences on specialized aspects of audio, will concentrate exclusively on the study and exploration of sound reinforcement and architectural acoustics. Conference Chairman Ted Uzzle and the conference committee have organized a program of in-

vited papers that promises to be an outstanding educational experience.

On the basis of their knowledge and experience, audio experts have been chosen to chair sessions on special disciplines within the field of sound reinforcement. Twelve sessions will provide an in-depth look at the technology, from earliest history to the latest developments and concepts.

Session topics and chairmen are as follows:

The History of Sound Reinforcement <i>Jesse Klapholz</i> 3730 Lankenau Rd. Philadelphia, PA 19131, USA	Sound Reinforcement for Mega-Events <i>Will Parry</i> Maryland Sound 4900 Wetheredsville Rd. Baltimore, MD 21207, USA	Computer Control of Sound Systems <i>Tom Roseberry</i> Innovative Electronic Designs, Inc. 9701 Taylorsville Rd. Louisville, KY 40299, USA
New Frontiers in TEF™ Applications (Memorial Session for Richard C. Heyser) <i>Eugene T. Patronis, Jr.</i> School of Physics Georgia Institute of Technology Atlanta, GA 30332-0430, USA	Computer-Aided Sound System Design <i>John Eargle</i> JME Consulting Corporation 7034 Macapa Drive Los Angeles, CA 90068, USA	Stereophonic Reinforcement Systems <i>Ron Streicher</i> Pacific Audio-Visual Enterprises 545 Cloverleaf Way Monrovia, CA 91016, USA
New Concepts in Loudspeaker Design <i>Mark Engebretson</i> 3404 Highway 79 Warner Springs, CA 92086, USA	New Concepts in Equalization <i>Chris Foreman</i> Panasonic Industrial Company 6550 Katella Ave. Cypress, CA 90630, USA	Sound Systems in Acoustically Difficult Rooms <i>Thomas Horrall</i> BBN Laboratories, Inc. 10 Fawcett St. Cambridge, MA 02238, USA
Sound Systems in the Live Theater <i>Rollins Brook</i> RB Systems 5717 Calvin St. Tarzana, CA 91356, USA	Electronic Enhancement of Reverberation <i>Christopher Jaffe</i> Jaffe Acoustics, Inc. 114A Washington St. Norwalk, CT 06854, USA	Sound Reinforcement in the Year 2000 <i>Cliff Henriksen</i> 413 E. Front St. Buchanan, MI 49107, USA

If your field is sound reinforcement, or if you want to learn more about this integral aspect of audio engineering, plan to attend the 6th Conference. For further information, contact:



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L&R: The Big Little Production Company

Here is a company that is using small-format gear to good advantage. They are up against some stiff competition — and winning a share of the national market.

THE WORLD OF BROADCAST PRODUCTION MUSIC IS DOMINATED by giants. They have built fortresses for themselves in cities such as Dallas, Memphis, San Diego, Boston, and of course New York.

But a challenge is being posed to these Goliath enterprises from the humble hills of Connecticut. Here, a band of modern day Davids (with slingshots in hand) are attempting to wrest a share of this competitive national market away from “the powers that be.” If this sounds a bit far-fetched, a visit to L&R Productions in East Hartford, Connecticut might convince you otherwise.

No, L&R is not another state-of-the-art studio. Rather, it is replete with narrow-gauge multi-tracks, mid-line signal processors and personal computers: exactly the kinds of gear you would expect to find in today’s electronic cottage. It is, however, L&R’s creative use of this equipment that has earned them much respect in their field. Sheer numbers give some reasons for their formidable reputation. Since opening their doors for business two years ago, L&R has churned out nearly 400 pieces of broadcast music, virtually flooding the regional market with their product and gaining clients as far west as New Mexico and California. If the slope of their sales curve means what I think it means, 1988 could very well be the year L&R becomes a national contender. How can five guys from Connecticut get this far on small-format gear? Let’s find out!

After spending an afternoon at L&R Productions, I am convinced that much of their success can be attributed to the spirit in which they conduct their business. Though everyone works long hours to complete productions on deadline, the simple joy of making music seems to always remain in their midst. United for a common goal, everyone in the shop functions as a highly valued team player. More than anything else, this atmosphere seems to be attributable to the management style of L&R’s founders.

STARTING UP

When Paul Lombardo and Tom Russo first conceived their company in June of 1985, it was nothing more than a telephone answering machine, a post office box and a multi-track sitting in Tom’s bedroom. Six months later they moved into their present facility, a cozy 1800 square foot

nook in a re-furnished turn-of-the-century textile mill. (Perched on the edge of a water fall, the mill still has a turbine which was formerly used to power the plant. Were that turbine restored to use, L&R would be a stellar example of the self-sufficient electronic cottage!)

As their business began to take off, Lombardo and Russo faced the inevitable temptation to build what they smilingly refer to as “the zoom room,” a world-class, multi-multi-track facility with all the right accoutrements. But this would have required taking on a massive debt, and after much deliberation they decided (for now, at least) to remain a small-format studio. Why? Simply to keep the business profitable to all concerned. Their priorities were to expand the base of personnel first, and let steady company growth ultimately build them “the zoom room” — without the pressures of paying off a large note. Above all, they didn’t want economic considerations forcing them to change the nature of the business they truly love: making commercial music for radio and television.

Rob Rainwater, L&R’s chief engineer (who also doubles as sales manager), articulated the company’s philosophy in this matter. Says Rainwater, “This is a music production company — not a recording studio. What I found when I was at other recording studios is you can have all the equipment in the world, but if you don’t have people keeping ‘em busy all the time, you’re not doing anything.”

Figure 1. In studio A we see Tom Russo (left), president of L&R; Paul Lombardo v.p.(center); and chief engineer, Rob Rainwater.



John Barilla's Electronic Cottage operation is in New York and it is based around an AKAI-1212.



Figure 2. The Tascam M-520 with the Fostex 4035 Synthesizer in studio A.

THE LINE UP

So L&R has opted to keep people real busy on their present equipment, and that they do. Each of the five main players at L&R is a capable producer/engineer in his own right. System protocols have been worked out so all of them can assume various roles where required. Owners Lombardo and Russo concentrate on the creative side and on client relations, but often engineer and perform as well.

Rainwater, who also wears many hats, is largely responsible for both the technical and marketing growth of the company. An excellent programmer, his voice editor/librarian software for the Yamaha DX-7 is currently being marketed by Voyetra Technologies (under the name "Sideman DTX"). The fourth member of the team is Jim Russo. A skilled percussionist and arranger, this younger Russo brother also functions as studio manager. The final team member is Marty Fegy, a respected composer of both orchestral and electronic music, whose work is frequently in performance throughout New England.

THE SYSTEM

Beyond these special areas of responsibility, any one of the five may find himself variously composing, arranging, producing or engineering a spot. Hence, the configuration of L&R's equipment needed to be made as user-friendly as possible. Rob Rainwater, who is largely responsible for the current system reflects, "The theme here is 'workable, moveable, manageable'...so you don't have to be a great technician to work here." For both audio and MIDI signal paths, both A and B rooms are wired parallel so that patch

Figure 3. Engineer Rob Rainwater at the Tascam MS-16 in studio A.



Figure 4. Composer Marty Fegy (left) confers with studio manager Jim Russo in the off-line production suite (B).

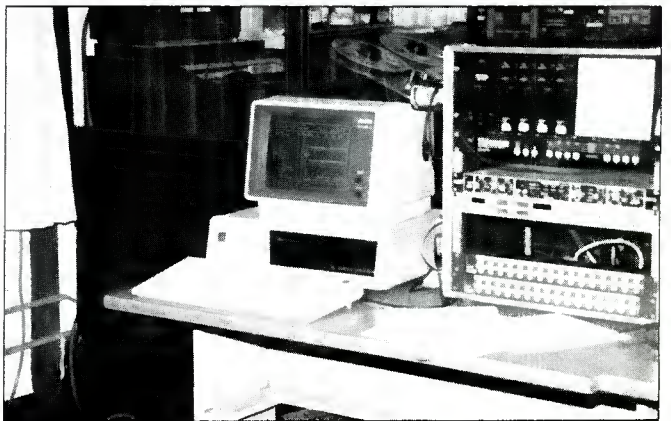
bays in either room are set up almost identically. Rainwater is working toward what one could call a *transparent system*: one which calls least attention to itself and does not inhibit the creative process. While some elite stand-alone recording systems (like Synclavier) are capable of doing this quite well, it is not really possible to achieve with combinations of modular equipment — each with different internal architecture. Still, just focusing in this direction can cause a user to make many beneficial steps toward clarifying the system he/she already has.

The main (A) room at L&R features the ever popular Tascam MS-16 multi-track, both 4- and 2-track mixdown on Otari 50/50s, a Tascam Model M-520 recording console and a respectable collection of outboard signal processors. (See equipment list). Monitoring is done through the standard triumvirate of UREI, Yamaha and Auratone speakers. So there is nothing unusual about this room, just some very acceptable "meat 'n' potatoes" type of gear. It all served L&R quite well until their business started to outstrip the available time in studio A (at that time, the only studio.)

PRODUCTION PROTOCOLS

Since the production of broadcast music (jingles, IDs, presentation themes, etc.) is largely electronic today, the logical quick-fix for L&R's increasing studio traffic was to prepare much of the material off-line, and use the multi-track room only when absolutely necessary. Staff composer Marty Fegy had been composing at home with an IBM compatible PC with sequencing and music notation soft-

Figure 5. The IBM XT with a hard drive and the auxiliary rack. The view is from studio B looking toward studio A.



ware and a rack of Yamaha TX modules. He advocated L&R installing an identical system in the studio, so that all he needed to transport was his data disk. When Rainwater joined L&R last year bringing with him his programming and technical experience, the idea of Studio B—a composition, pre-production, demo room—was born.

In this off-line suite, one, two or more composers or engineers (depending on what hat they are wearing) can be productive simultaneously, while the on-line Studio A is humming away laying tracks or mixing. Because there are audio tie lines, compatible computers in both rooms, as well as central MIDI patching, it is possible not only for each room to work independently, but for *either* room to assume *control* of the other, when necessary. Even a redundant remote control for the 16-track machine has been supplied in Studio B, so that on-line recording can be achieved without having to re-configure the system.

One very useful feature of the Tascam M-520 board is the multitude of possible independent signal paths that are designed into it. (Undoubtedly, this finds a major application in TV mixing where various separate mixes may be required). For L&R this will allow for simultaneous processing of material from the A and B rooms without interference—a tremendous asset when time is essential. (Of course, the B room does have its own mixer which can function independently or in conjunction with the 520).

EFFICIENCY

From experience, L&R has found that broadcast production music has some very well-defined parameters that allow much of the sound generating equipment to be normalised through, so that repeated setup is no longer necessary. The kind of open-ended experimentation that we all like to do on record sessions is simply not relevant here. Those searches for the “perfect” sound, while many times fruitful, are just as frequently fruitless. At L&R, experimentation is not stifled, but once a satisfactory setting is found, it’s “remembered” by the system for a quick setup. For example, the Neumann U-87 that Paul Lombardo uses is always patched into a certain channel. Is he ready to sing? Then lift the fader to “7,” punch in the pre-set EQ and you are ready to record. It is likely the mic trim has already been set, the compressor already patched. Different vocalist today? No problem, just patch over to a pre-set graphic in place of the console EQ, adjust the trim and press record.

Does this all sound cold and artless? Really it isn’t. Due to the special requirements of the medium (for example, thickly textured group vocals), the same bevy of vocalists

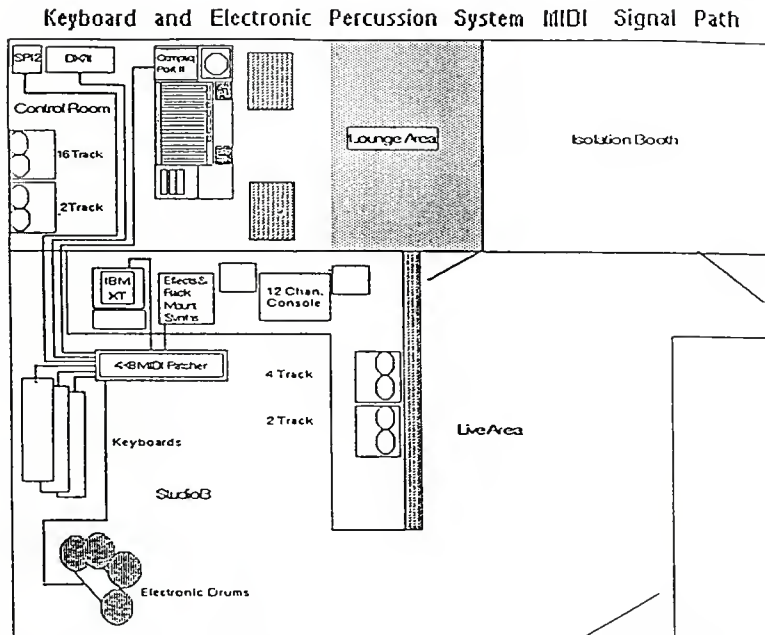


Figure 6. The studio floor plan.

gets called time and again. Why? Because everyone knows everyone: assets, limitations, everything! Now it only takes minutes instead of hours to get the quintessential vocal sound from that group—and saving time turns into dollars for everybody.

STATION IDs

Speaking of smokin’ vocal sounds, everyone knows how hot the vocals sound on radio station IDs. L&R had been doing quite a few of them lately, so I asked them to share with *db*’s readership some tips on getting a big sound—particularly on small format equipment with limited tracks. After a

period of experimentation, the company settled on this method:

- 1) Sequence the musical bed in an off-line facility.
- 2) Record onto the 16-track: stereo drums, stereo keyboards, mono-bass and SMPTE. Hence, the majority of tracks remain open for marathon vocal session.
- 3) Vocal lines (usually in three parts) are previously charted by music notation software and a synthesized “guide” track for vocalists is laid down.
- 4) Three excellent vocalists (usually two females and a male), gathered around one microphone, sing (together) one part at a time in unison or octaves.
- 5) Track each part three times (3 parts x 3 tracks x 3 singers = 27 voices.) If necessary, consolidate tracks down to the three basic parts, and re-synthesize stereo via signal processing.
- 6) Add guitars or other live instruments on remaining tracks.

While mic’ing this session, moving the three singers to various positions around a single mic can disguise their individuality and maximize the strength of their ranges. Additionally, this movement will bring about timbral shifts which give the illusion of a much larger ensemble.

Carefully followed, the above technique will work every time with, of course, the following proviso: “All technical and electronic aspects aside,” says Tom Russo, “one of the keys to getting a great vocal sound is having the right singers; to be able to come in having the right diction, phrasing and feel, and be able to blend well together, to stand out when necessary, maintain their individuality, yet create the cohesiveness for a radio packaged sound. It takes a pro; there’s no question about it!”

REFLECTIONS ON MIXING

I queried them further, about the techniques of mixing for their speciality. While I never quite thought about it too deeply prior to this, it occurred to me that in terms of learning his/her craft through immediate feedback, nothing beats mixing broadcast production music. Let’s say for ex-

ample that you mix an album. Unless it's a hit, you will probably never hear it in its entirety over the air-waves. To correlate a lot of information about your mixes will take years. Not so with these people who do one minute commercials and other such things; *they* get to hear it on radio or TV within a few days after they mix it. They learn their craft at an accelerated rate.

I asked Paul Lombardo to share some of the mixing methodology he's developed over the years. "Get the sounds on the big boys," he said, pointing to the UREIs, "then mix at low volumes on the small speakers. The goal is to keep those "highs" in proportion, 'cause once they come out on the radio—once they compress it and cart it—it has a different sound." He went on to say that a frequency skewed mix—though interesting on a studio monitor—may be exaggerated by the broadcast process and turn aural titillation into ear fatigue or worse. Lombardo's final advice: "Mix big on the low-end. You definitely want the clarity on top, but if there isn't an extra dollop of low-end, commercials can sound thin and harsh over the air." Studio manager Jim Russo noted that there really isn't any substitute for trial and error and careful monitoring of the air-waves, "Fortunately, we all have collective memory here. We all listen and store the information for further projects."

No one just walks into a viable business today—especially in a highly competitive marketplace such as broadcast production music.

MAKING IT HAPPEN

As I observed the enthusiasm and synergy between the entire L&R team, I reflected on how this "esprit de corps" is possible only because Lombardo and Russo found a way to turn their dreams into green dollars and common sense. No one just walks into a viable business today—especially in a highly competitive marketplace such as broadcast production music. It takes boldness, persistence, and above all, vision. Tom Russo comments on his experience building L&R Productions over the past few years, "There are two sides to the music business. Music and business. If it wasn't for the business side, we wouldn't be here right now. We found out pretty quick that most of our competition was not in Connecticut. So our concept was, 'Hey, we're here in your back yard. We'll be over in 10 or 20 minutes.' And we just started taking good care of our clients."

"Sure, we needed to keep parallel jobs for a while. No bank will fund such a venture. You have to build slowly. Customer service is important, going the extra mile even when the client does not require it."

Some techniques L&R used to market their services sound so commonplace, they seem too normal for the music industry. Yellowpage ads, for instance. According to Russo, his Yellow page ad "paid for itself after one month in print." Other techniques seem pretty esoteric. For example, L&R is now pursuing the "sports music" market.

Having done some successful themes for the Hartford Whalers, Boston Red Sox and Bruins, they theorized that a large specialty market was about to emerge. To test their hypothesis they recorded a few pieces and flew down to the National Association of Baseball Leagues convention in Dallas, Texas and came away with 8 new clients!

From the esoteric to the commonplace: no stone can be left unturned when starting a new business. In L&R Productions, small-format studios everywhere have a shining example of what can happen when talent, persistence and cooperation gather under one roof. db

EQUIPMENT LIST

Recorders and mixing console:

Tascam MS-16 16-track
Otari MX5050 QII 4-track
Otari MX5050 BII 2-track
Tascam M-520 mixing console

Monitors:

UREI 809 Biradial studio monitors
Yamaha NX10

Auratones

Effects and Outboard Gear:

Yamaha Rev 7 digital reverb
Roland SRV 2000 digital reverb
Yamaha R1000 digital reverb
Yamaha D1500 digital delay
Yamaha SPX90II
Loft 450EM analog delay/flanger
dbx 166 gate/compressor
Symetrics 501 compressor
Loft parametric eq
dbx Type I noise reduction
Shultz Studio Rockman

Microphones:

Neumann U87
AKG 414, 451, 460
Crown PZM
Shure SM57
E-V RE15

Instruments, etc:

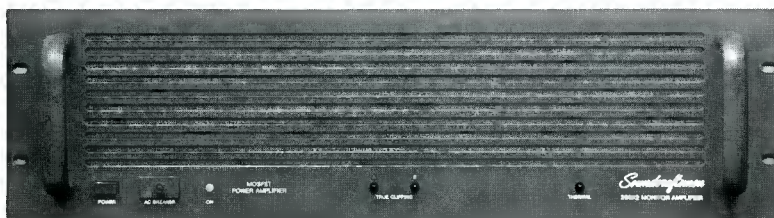
Yamaha DX7, TX816, DX7II, TX81Z
Emulator EMAX sampler
Roland JX8P
Emulator SP12
Emulator Drumulator
LinnDrum
IBM PC/XT with hard disk
Personal Composer
Sequencer Plus MKIII
Patch Master Plus
Sideman DTX
12 guitars
6 bass guitars
Boogie guitar amps
Tama drums
Roland Octapad electronic drums
Synchronizer:
Fostex 4035

Lab Report

Soundcraftsmen Model 200X2 Stereo Power Amplifier

GENERAL INFORMATION

Soundcraftsmen has always managed to come up with extremely reliable professional and consumer audio amplifier equipment and their new X2 series of amplifiers maintains that tradition. The most powerful model in that series is their 450X2, rated at 300 watts per channel into 4 ohms and 205 watts per channel into 8 ohms, both channels driven, from 20 Hz to 20 kHz at less than 0.05 percent THD. All of these amplifiers can be operated in a bridged mode as well as in stereo. Stereo or mono mode selection is available by means of a rear panel switch; no internal wiring or jumpers necessary. Mono operation of the highest powered unit yields 630 watts.



The particular unit we tested boasts a more modest power rating of 125 watts per channel at 8 ohms; 190 watts per channel at 4 ohms, and 380 watts in bridged/mono mode at 8 ohms. In fact, our

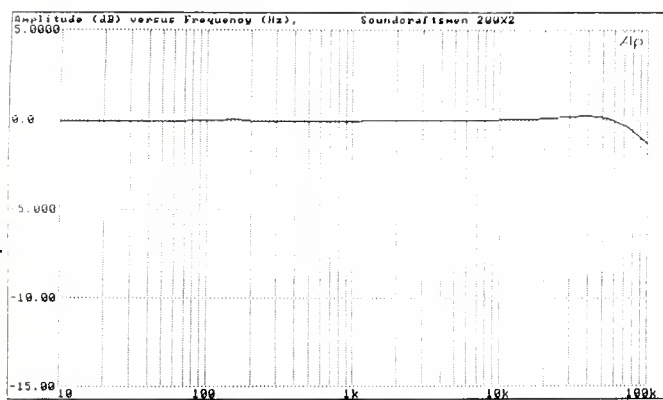
sample did a good deal better than that and also exhibited a high level of dynamic headroom. That means that for short term peaks (such as those that occur during actual program material) the amplifier is capable of delivering nearly twice its rated power.

The 200X2 is available either with two rows of LEDs used as power output meters (Model 200X2M) or without. Our sample came without the meter option. The amplifier uses MOS-FET output stages which are generally credited with providing high current capability and the ability to drive low impedance loads. The 200X2 is equipped with internal protection circuitry against possible shorted, mismatched or open circuits.

An extremely versatile input section permits the use of XLR or 1/4-inch phone plug connectors as well as connection via a 5-terminal barrier strip. All inputs are wired in parallel for convenient amp patching and all accept either balanced or unbalanced inputs. Internal construction is semi-modular, making servicing relatively easy. The heavy-

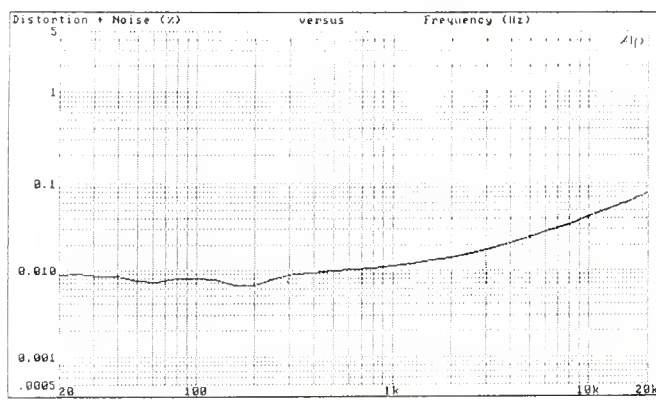
In fact, our sample did a good deal better than that and also exhibited a high level of dynamic headroom.

Figure 1.

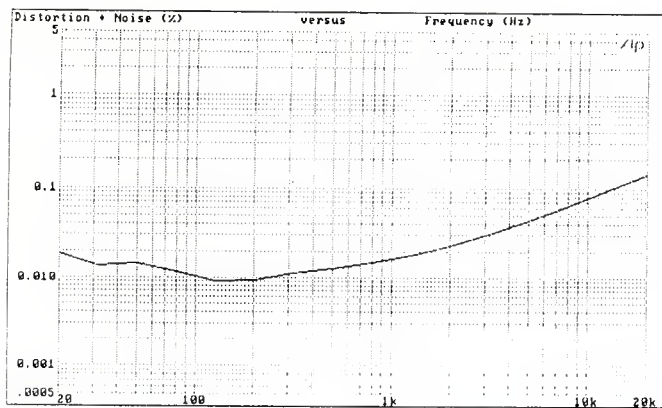


Frequency Response, Soundcraftsmen 200X2 Amplifier

Figure 2(A).

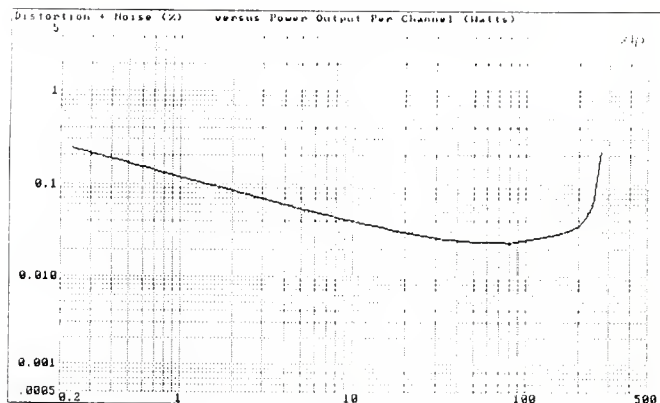


Distortion + Noise versus Frequency at Rated Power (125 watts, 8 ohms) Soundcraftsmen 200X2



Distortion + Noise versus Frequency at Rated Power (190 watts, 4 ohms) Soundcraftsmen 200X2

Figure 2(B).



Harmonic Distortion plus Noise versus Power Output per Channel (4 ohm loads, 1 kHz), Soundcraftsmen 200X2

Figure 3(B).

duty front panel is equipped with carrying handles and is configured for standard 19-inch rack mounting.

CONTROL LAYOUT

In the meterless version that we tested, the all-black front panel is equipped with a pair of "true clipping" LED indicators along the lower edge of the panel and near its center. Further to the right is an LED labeled "thermal" that is supposed to illuminate if operating temperatures of the output stages become excessive. This indicator never came on during all of our testing; not even during the one hour pre-conditioning test at 1/3 rated power that is designed to stress the amplifier to near-maximum internal power dissipation.

**Dynamic headroom
measured an impressively
high 2.9 dB with 8 ohm
loads.**

The only controls found on the front panel are the main power switch and an AC circuit-breaker reset pushbutton. We should note that never, during all of our bench tests, was it necessary to push the reset button, either. Bear in

mind that during those tests, the amplifier was often called upon to deliver its rated power or even more for extended periods of time—a condition that is not likely to occur during actual use of the amplifier.

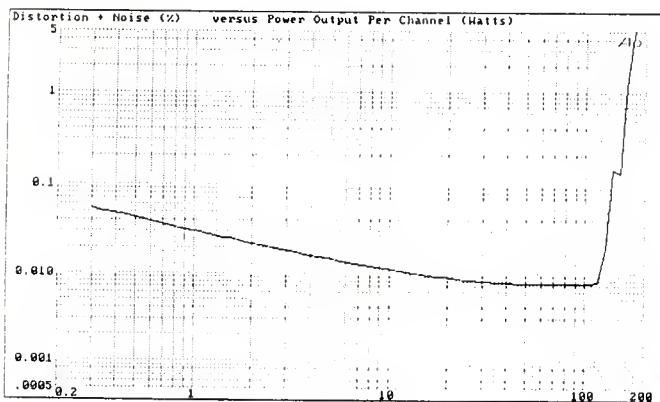
Massive finned heat sink structures extend at the rear of the amplifier and the output devices are attached to these heat sinks for maximum heat transfer. Between the projecting heat sink structures, centered on the rear panel, are a pair of female XLR input connectors, a pair of 1/4-inch phone jack inputs and a five-terminal barrier strip. The center terminal of this strip is a ground terminal while outer terminals are labeled "+" and "-" for each channel, so that even if connection is made via this barrier strip, it is still possible to choose a balanced or unbalanced input configuration. On either side of the barrier strip is a rotary input level control, calibrated from +6 dB to minus "infinity." The "0 dB" calibration mark corresponds to an amplifier voltage gain of +26 dB.

A slide switch beneath the terminal strip is used to select stereo or mono (bridged) operation. Color coded 5-way binding posts to the right of center on the rear panel are used to connect the amplifier to its speaker loads. In the bridged mode, a single load is connected between the two binding posts marked "+."

LABORATORY MEASUREMENTS

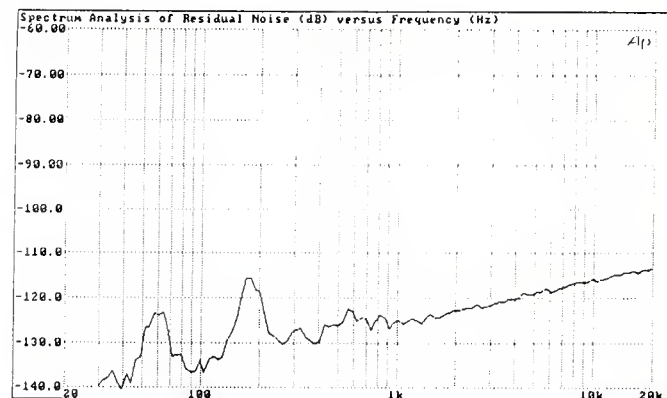
The VITAL STATISTICS chart at the end of this report summarizes all of the bench test measurements performed on this amplifier and compares these results with Soundcraftsmen's own published performance specifications.

Figure 3(A).



— Harmonic Distortion plus Noise versus Power Output per Channel (8 ohm loads, 1 kHz), Soundcraftsmen 200X2 —

Figure 4.



Signal-to-Noise analysis as a function of frequency, Soundcraftsmen 200X2

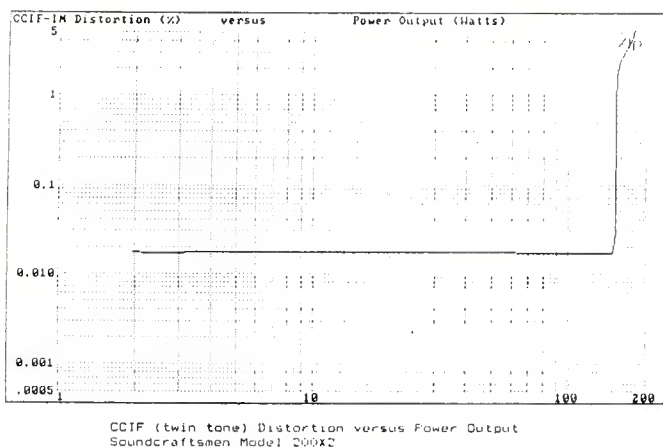


Figure 5.

Our laboratory has recently been equipped with new, computer-driven test hardware that provides extremely accurate and rapid test results that can be printed out directly by our graphics-capable printer. The graphs reproduced in this report, therefore, were generated by this system in real time, as the amplifier was subjected to its various tests.

Figure 1 is a plot of amplitude versus frequency, extending from 20 Hz to 100 kHz. The vertical scale is in dB and, as you can see, response was virtually flat out to 60 kHz. Even at 100 kHz, response was down only 1.5 dB.

Figures 2A and 2B are plots of harmonic distortion versus frequency, with output held constant at rated output (125 watts per channel, 8 ohm loads in Figure 2A; 190 watts per channel, 4 ohm loads in Figure 2B.) Over much of the useful frequency range, when coupled to 8 ohm loads, the amplifier's THD plus noise was well below the rated 0.05 percent specified (it measured a mere 0.011 percent at 1 kHz.) At 20 kHz, THD plus noise did rise a bit above the rated value, measuring 0.078 percent, but that's hardly worth quibbling about.

As shown in Figure 2B, when connected to 4-ohm loads, and delivering 190 watts of power per channel, THD plus noise at 1 kHz measured 0.017 percent — again, well below the rated value of 0.05 percent. At 20 kHz, however, THD + noise rose to a level of 0.15 percent — still nothing to get terribly excited about.

Figures 3A and 3B are plots of total harmonic distortion plus noise versus power output level, for 8 and 4 ohm loads respectively, using a 1 kHz test signal. The rated THD + Noise level of 0.05 percent was maintained from 300 mW up to 138 watts per channel, both channels driven, in the case of 8 ohm loads (Figure 3A) and to as high as 220 watts per channel in the case of 4 ohm loads (Figure 3B). The seemingly higher THD plus noise at low power levels in the graph of Figure 3B is primarily due to the relative noise levels referred to low power level rather than to actual harmonic distortion.

Soundcraftsmen's published specification of signal-to-noise ratio for this amplifier is referenced to rated output and is given as -105 dB. Measured that way, we obtained a S/N ratio of -107 dB. However, the EIA standard for amplifier measurements calls for the measurement of S/N to be made relative to a 1 watt output, using a fixed input of 0.5 volts (and adjusting the input level controls to provide the required 1 watt output.) Measured in that manner, we read a S/N ratio of -84.7 dB (A-weighted.) While that is a "smaller" dB figure, we would hasten to add that it is an ex-

cellent S/N figure for this method of measurement. To further explore the nature of the residual noise of the amplifier, we ran a spectrum analysis of the residual noise components of the amplifier, from 20 Hz to 20 kHz. Results are shown in Figure 4. Note that the major contributory factors to the residual noise are a 60 Hz and 180 Hz component obviously related to the power supply and the power line frequency. Even at that, the 180 Hz component (the worst of the two) is still down approximately 115 dB below rated output!

SMPTE Intermodulation Distortion measured only 0.01474 percent on one channel and 0.01450 percent on the other channel at rated output, using 8-ohm loads. CCIF (twin tone) intermodulation distortion, often regarded as a more significant type of IM than SMPTE-IM, was measured over a power output range from 2 watts per channel to beyond overload and clipping, and these results are shown in the graph of Figure 5. CCIF-IM (using 18 kHz and 19 kHz test tones) remained below 0.02 percent until obvious clipping occurred at a level of approximately 150 watts per channel.

Dynamic headroom measured an impressively high 2.9 dB with 8 ohm loads. That means that for short term musical peaks, the amplifier can deliver as much as 243.7 watts per channel before the onset of noticeable clipping! Damping factor, referred to 8 ohms, measured 220, as against 200 claimed by Soundcraftsmen.

**the amplifier is so rugged
that it is likely to continue to
provide excellent service
even under the most trying
and demanding circum-
stances.**

COMMENTS

In judging an amplifier these days, whether the amplifier is designed for professional use or for "high fidelity" applications, I consider sound quality and accuracy to be of paramount importance. Because Soundcraftsmen's amplifiers are popular with both audiophiles and in sound reinforcement applications, I have always found that their "pro" amplifiers exhibit the same sonic excellence and transparency that is the hallmark of their "hi fi" units. But a "pro" amplifier needs to deliver more than just good, accurate sound. It must often stand up to the rigors of less than gentle transportation, frequent overload and high power demands and, yes, even misuse. If there is one thing that can be said about the Soundcraftsmen 200X2 it is that the amplifier is so rugged that it is likely to continue to provide excellent service even under the most trying and demanding circumstances. As indicated earlier, we never had to reset the circuit breaker during our tests, and despite the fact that we ran the amplifier at high power output levels, we never saw the thermal indicator light go on. That says a lot about how conservatively this amplifier has been designed. Whether you buy the 200X2M version, with LED meters, or the 200X2 version that we tested, these Soundcraftsmen pro power amplifiers will give you your money's worth, and then some.

VITAL STATISTICS

SPECIFICATION	MFR'S CLAIM	db MEASURED
Power Output, 20 Hz to 20 kHz		
8 ohm Loads	125 W/channel	138 W/channel
4 ohm Loads	190 W/channel	220 W/channel
8 ohms, bridged	380 Watts	405 Watts
Frequency Response	20 Hz-20kHz, ± 0.1 dB Confirmed	
Rated THD	0.05%	0.078% (See Text)
Hum and Noise	-105 dB	-107 dB (See Text)
Slew Rate	40 V/microsecond	More than 50V/usec
Rise Time	2.2 microseconds	2.0 microseconds
Input Sensitivity	1.0 Volts	0.72 Volts
Input Impedance		
Balanced	22k-Ohms	Confirmed
Unbalanced	32k-Ohms	Confirmed
SMPTE-IM Distortion (8 ohms)	N/A	0.015% @ 125 W.
CCIF-IM Distortion (8 ohms)	N/A	0.006% @ 125 W.
Dynamic Headroom (8 ohms)	N/A	2.9 dB.
Damping Factor, 8 ohms	200	220
Power Requirements	100-125 VAC, 50/60 Hz	
Dimensions:	19" std. rack, 5.25" h, 10.5" behind mounting surface.	
Weight:	28 lbs.	Confirmed
Suggested Price:		
Without meters	\$699.00	
With meters	\$799.00	

Circle 30 on Reader Service Card

Buyer's Guide: Power Amplifiers

Introduction to the Charts

We've tried to make the charts of amplifiers as self-explanatory as possible, with slanting headlines on each column that explain what we wanted to show you.

These charts represent entirely what each of the respective manufacturers have sent us in response to our (sometime repeated) requests. You will also see that there are numbers of blank sections within the charts. If they don't have a specification available, we can't list it. But note that many do not have anything under the Features column. This column is where we have invited each manufacturer to state, in as few words as possible, what is special about the product. You can safely assume, then, that when this column is blank, it is because the manufacturers told us nothing.

Note also that we ask for amplifier continuous power not only at the traditional 8 and 4 ohm resistive loads, but also at 2 ohms. As you know, when you parallel speakers, the load is halved. Accordingly, in the real worlds of studio monitors and headphone lines, and the even more real world of performance and stadium systems, effective loads back to an amplifier can well be 2 or 3 ohms. Since modern solid-state amplifiers can handle such loads successfully, we ask each manufacturer for this specification. Note that not all give it. It's, therefore, safe to assume that if it is missing, the amplifier may not be reliable at low loads.

Distortion at normal and full power ratings is also specified. While many amplifiers today can boast of almost vanishing distortion, remember that if you will be pushing an amplifier hard up against its rated power and beyond, distortion will then be rising rapidly. No audio product is really made to be abused, and amplifiers are no exception.

One group of important specifications deals with dimensions and weights. Amplifiers, particularly high-power ones, are not lightweights. A few racks can have weights adding up rapidly.

Finally, the price. What we have asked each manufacturer for is the suggested retail price. Different retail dealers establish their own.

On to the charts...

A Buyer's Guide To Power Amplifiers

Model	Number of Channels	Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-KHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, \$	Features
ALTEC LANSING CORPORATION															
1590E	1	200	20-20k					1	20-20k 1	0.8	10.5 19 8.25	41		\$1612.00	
1269	2	120	200	20-20k				.05	20-20k .25	.775	3.25 19 14.75	31		\$1100.00	
1268	2	60	100	20-20k				.05	20-20k .25	.775	3.25 19 10	27		\$900.00	
1270B	2	220	400	20-20k				.05	20-20k 3	.775	3.25 19 15.25	52		\$1552.00	
2204A	4	75		20-20k				.25	20-20k 1	.775	5.25 19 12.75	36.5		\$1576.00	4 ch. incremental power system. These 75W channels can be paralleled and/or bridged.
9444A	2	200	300	20-20k				.05	20-20k 1	.775	5.25 19 12.75	39		\$900.00	Special heatsink achieves a temperature differential of 7.4C, which is lower than others tested.
1593C	1	50		20-20k				1	20-20k 1	.8	5.25 19 7.38	23		\$1056.00	
1594C	1	100		20-20k				1	20-20k 1	.8	7 19 8.5	35		\$1236.00	
AMR (Audio Media Research) See our ad on Cover III															
PMA200	2	100						.008	20-20k .2	1	5.25 19 12.125	29		\$399.50	
PMA70+	2	30	30					.01	20-20k 1	.775	5.25 7 8.25	10		\$179.50	100W per channel instantaneous output power (dyn. headroom 10dB at 4 ohms), rear mounted controls.

Model	Number of Channels				Cont. power/channel at 8 ohms, all channels driven				Cont. power/channel at 4 ohms, all channels driven				Cont. power/channel at 2 ohms, all channels driven				Power Bandwidth	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, \$	Features
BIAMP SYSTEMS																											
T500	2	150	240	300												.05	.05	20-20k .5	1.4	5.4 19 10	35	\$1199.00				The T series are MOSFET power amplifiers. High reliability in uninhibited 2-ohm operation.	
T1000	2	310	480	680												.04	.05	20-20k	2	7 14	48	\$1599.00					
XA100	2	35	50													.05	.05	20-20k .5	1.65	1.75 19 11.75	13.5	\$499.00				The XA series are MOSFET power amplifiers. MOSFET devices conduct less current as temperature increases.	
XA300	2	100	150													.1	.07	20-20k .5	1.85	3.5 19 15.5	21	\$649.00				Same as unit above.	
XA600	2	200	300													.1	.07	20-20k .5	2.85	3.25 19 15.5	24	\$849.00					
XA1000	2	300	500													.1	.1	20-20k .5	2.45	5.25 19 12	35	\$1199.00					
BOULDER AMPLIFIERS																											
500	2	500	250		0.015- 100k	.001	.003	0.0	.005	20-20k 0.04	1.75	7 19 15.5	51	\$2875.00													Two-stage design amplifier with stereo mono switch and balanced inputs.
CARVER CORPORATION																											
PM2.0t	2	465	625		20- 20k	.1	.5	.1	.5	20-20k .5	2.16	3.5 19 12.25	10	\$1660.00													Switchable clipping eliminator, balanced XLR and TRS inputs, dual speed fan, six-way protection.
PM1.5	2	450	600		20- 20k	.1	.5	.1	.5	20-20k .5	1.5	3.5 19 12.06	21	\$1160.00													70V direct line ability, balanced XLR/TRS inputs, power-up sequencer, six-way protection.
PM350	2	350	450	450	20- 20k	.1	.5	.1	.5	20-20k .5	1.5	3.5 19 11.56	21	\$940.00													Switchable clipping eliminator, bridged mono operation, optional on-board signal processing.
PM175	2	175	250	300	20- 20k	.1	.5	.1	.5	20-20k .5	1.5	3.5 19 11.56	19	\$720.00													Balanced XLR/TRS inputs, bridged mono operation, 6-way protection, optional on board sig. processing.
CARVIN CORPORATION See our ad on page 17																											
FET2000	2	400	600	1000	20- 20k	.005	.05	.005	.05	20-20k .5	1	5.25 19 13	55	\$995.00													High current MOSFET design, clip LED, signal LED, hi/lo cut filters compr., bridgeable, XLR/.25 inch.
FET900	2	200	300	450	20- 20k	.005	.05	.005	.05	20-20k .5	1	5.25 19 10	35	\$599.00													Same as above.
FET400	2	100	200		20- 20k	.006	.05	.006	.05	20-20k .5	1	5.25 19 10	28	\$499.00													Same as above.
DCM301	1	100	150	300	20- 20k	.01	.05	.01	.05	20-20k	1.5	5.25 19 10	35	\$359.00													Built-in nine band graphic EQ with bypass, clip LED, XLR and .25 inch balanced inputs.

Model	Number of Channels			Cont.power/channel at 8 ohms, all channels driven		Cont.power/channel at 4 ohms, all channels driven		Cont.power/channel at 2 ohms, all channels driven		Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, \$	Features							
CREST AUDIO																											
1001A	2	40	70	20-20k	.001	.01	.004	.01	1-50k	.47	1.75 19 10.5	18	\$660.00	Convection cooled, headphone out on the front panel.													
1501A	2	90	150	20-20k	.001	.01	.004	.01	1-50k	.63	1.75 19 10.5	19	\$720.00	Convection cooled, headphone out on the front panel.													
FA800	2	220	360	20-20k	.001	.01	.01	.05	20-20k	1	3.5 19 13	32	\$840.00	Uses a rear to front tunnel cooling design.													
PL400	2	290	450	20-20k	.001	.01	.01	.05	20-20k	1.17	3.5 19 13	38	\$1100.00	Includes a variable speed DC fan.													
3000/01	2	240	430	640 20-20k	.001	.01	.003	.008	1-50k	1.06 2.12	5.5 19 11.5	46	\$1589.00	Models 3000 and 4000 include large LED meters adding \$200 to the cost of the 3001 or 4001.													
4000/01	2	325	550	800 20-20k	.001	.01	.003	.008	1-50k	1.2 2.4	5.5 19 11.5	58	\$2190.00														
8001	2	750	1200	1400 20-20k	.001	.01	.007	.01	10-50k	1.75	5.5 19	80	\$2990.0														
CROWN INTERNATIONAL																											
Macro Tech 600	2	235	325	340 35k	.05	.05	.001	.05	20-20k .1	.77	3.5 19 16	43	\$1199.00														
Macro Tech 1200	2	320	465	600 35k	.05	.05	.001	.05	20-20k .1	.77	3.5 19 16	48	\$1499.00														
Macro Tech 2400	2	525	800	1200 35k	.05	.05	.001	.05	20-20k .1	.77	3.5 19 16	60	\$1899.00														
D-75	2	35	45	60 35k	.01	.01	.001	.05	20-20k .1	.81	2.7 19 9	13	\$524.00														
D150A Series 2	2	95	150	108 35k	.01	.01	.001	.05	1-20k .1	1.29	5.2 19 8.7	27	\$749.00														
DC300A Series 2	2	175	285	320 35k	.01	.01	.001	.05	0-20k .1	1.75	7 19 9.7	45	\$1049.00														
PS-200	2	95	135	172 35k	.01	.01	.001	.05	0-20k .1	1.3	5.2 19 10.2	35	\$819.00														
dbx																											
BX1	4	100	200	350 20-20k	.05	.05	.05	.05	20-20k .075	1	7 19 25	85	\$3700.00														
ELECTRO-VOICE, INC. See our ad on page 3																											
7300	2	200	300	20-20k				.01	10-100k -3	.902	5.25 19 12.75	39	\$958.00	Designed especially for musical instrument applications, has XLR and phone jack inputs.													
AP2600	2	200	300	20-20k				.01	10-100k -3	.902	5.25 19 12.75	39	\$958.00	Has barrier strip in/outs, controls on rear, for fixed installations, pro, motion picture.													
AP2600SA	2	200	300	20-20k				.01	10-100k -3	.902	5.25 19 12.75	39	\$958.00	Designed for fixed installations and pro audio applications.													

Model	Number of Channels			Cont. power/channel at 8 ohms, all channels driven	Cont. power/channel at 4 ohms, all channels driven	Cont. power/channel at 2 ohms, all channels driven	Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, \$	Features
												+/- dB					
DAVID HAFLER COMPANY																	
P500	2	255	400	20-20k	.007	.007	.025	.025	4-160k 3	1.55	7 19 13	53	\$1095.00	Front panel controls, clip ind., mono switch, signal ind., balanced line input.			
P505	2	255	400	20-20k	.007	.007	.025	.025	4-160k 3	2.35	7 19 13	48	\$995.00	Rear panel controls, mono switch, optional 70V line transformers.			
P230	2	115	175	20-20k	.005	.005	.02	.02	2-160k 3	1.55	5.25 19 10.5	28	\$560.00	Rear panel controls, mono switch, optional balanced line input and 70V line transformers.			
P125	2	62		20-20k	.005	.005	.009	.009	2-200k 3	1.1	3.5 19 9	19	\$450.00	Rear panel controls, mono switch, optional balanced line input.			
HILL AUDIO, INC.																	
DX300	2	200		20-20k		.002		.003	20-20k .5	1.55	3.5 19 8.5	16	\$830.00	DC servo controlled fans, relay protection for amp and load, .25 inch and XLR balanced inputs, 2 rack space amps have barrier strip for connections, groundlift capabilities to separate AC and audio ground, input sensitivity controls on front of amps, two LEDs on front panel to indicate signal presence and peak, all models use toroidal transformers for the power supply.			
DX800	2	250	400	20-20k		.002		.003	20-20k .5	1.55	3.5 19 13	29	\$1049.00				
DX1500	2	300	500	750 20-20k		.002		.003	20-20k .5	1.55	3.5 19 13	34	\$1299.00				
DX1000A	2	500	800	20-20k		.002		.003	20-20k .5	1.55	5.25 19 12	40	\$2049.00				
DX3000	2	550	900	1500 20-20k		.002		.003	20-20k .5	1.55	5.25 19 18	80	\$2749.00				
HM ELECTRONICS, INC.																	
PA120	2	60	100	20-20k				.01	20-50k 3		3.5 19 8.75	13	\$476.00	Lightweight, rugged, 2-color LED front panel tells if mono (double power) or stereo.			
INDUSTRIAL RESEARCH PRODUCTS, INC.																	
DH4020	2	100	140	20-20k				.1	20-20k 1	1	1.75 19 14	13.5	\$1012.00	Passive cooling, 100kHz switching power supply, MOSFET design, thermal prot., mute on/off, clip ind.			
JBL PROFESSIONAL																	
6215	2	35	45	20-20k	.1	.1	.1	.1	20-20k 1	1.1	1.75 19 9	10.5	\$597.00	The following features apply to models 6215, 6230, 6260, 6290: Active, balanced bridging input circuitry; full complementary driver and output circuitry; low transient intermodulation distortion; rugged, road-worthy construction; individual stepped gain controls; XLR, phone jack and barrier strip input connectors; heavy duty 5-way output binding posts; rear panel switch for bridged, dual mono, or stereo operation.			
6230	2	75	150	20-20k	.1	.1	.1	.1	20-20k 1	1.1	5.25 19 11	26.3	\$639.00				
6260	2	150	300	20-20k	.1	.1	.1	.1	20-20k 1	1.1	7 19 11	44.5	\$897.00				
6290	2	300	600	20-20k	.1	.1	.1	.1	20-20k 1	1.1	7 19 14	63	\$1347.00				
6211	2	40		20-20k	.01		.01		20-20k 1		8 8.5 2.75	6.5	\$312.00	Converts series 4400 or any other 8 ohm speaker system into a fully contained sound system. The 6211 features a built-in microphone preamp, a mic/line switch and a low-cut filter switch.			
6210	2	40		20-20k	.01		.01		20-20k 1		8 8.5 2.75	6.5	\$279.00				

Model	Number of Channels				Cont.power/channel at 8 ohms,all channels driven		Cont.power/channel at 4 ohms,all channels driven		Cont.power/channel at 2 ohms,all channels driven		Power Bandwidth	Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W	+/- dB	Sensitivity for full output,V	Dimensions, H/W/D, in.	Weight, lbs.	Price, \$	Features
PEAVEY ELECTRONICS CORPORATION																							
Deca 528	2	210	250		20-20k								.1			1	1.75 19 14	12	\$649.00		Compression, compression defeat, output protection, linear phase response, 90% power transfer.		
Deca 724	2		350		20-20k								.1			1	3.5 19 18	37	\$999.50		Same as above.		
Deca 1200	2		600		20-20k								.15			1.3	3.5 19 18	37	\$1399.50		Same as above		
M4000	2		200		20-20k								.05			1	5.25 19 12.38	35	\$579.40		Automatic 2-speed fan, independent channel thermal/ fault protection, compr., LED ind.		
M7000	2	200	350		20-20k								.05			1.4	5.25 19 16.5	47	\$579.50		Compression, 2-speed fan, thermal and fault protection, triac speaker protection system, LED ind.		
CS800	2	230	400		20-20k								.1	5-40k 1		1.4	7.38 19 13.19	54	\$799.50		Compression, 2-speed fan, calibrated level controls, bridging, electronic crossover islands.		
CS900	2	250	450	350	20-20k								.05			1.4	5.63 19 16.38	53	\$949.50		Automatic 2-speed fan, thermal/ fault protection, triac speaker protection, XLR/dual phone inputs.		
CS1200	2	350	600	600	20-								.05			1.4	7.5 18	71	\$1299.50		Compression, 2-speed fan, two in-lands, thermal/fault protection.		
QSC AUDIO PRODUCTS																							
1080	2	35	50		40k						.01	.01	.01	20-20k 1		.83	1.75 19 7	12	\$488.00				
1400	2	200	300		40k						.01	.01	.01	20-20k 1		1	5.25 19 9.5	34	\$768.00				
1700	2	325	500		40k						.01	.01	.01	20-20k 1		1	7 19 10.8	57	\$1098.00				
MX 1500	2	330	500		40k						.01	.01	.01	20-20k 1		1	3.5 19 17.9	47	\$998.00				
MX 2000	2	375	625		40k						.01	.01	.01	20-20k 1		1	5.25 19 15.9	75					
3200	2	110	140		40k						.01	.01	.01	20-20k 1		1	1.75 19 14.6	26	\$958.00				
3500	2	300	450		40k						.01	.01	.01	20-20k 1		1	3.5 19 15.9	50	\$1488.00				
3800	2	375	600		40k						.01	.01	.01	20-20k 1		1	5.25 19 15.9	75	\$1958.00				
RAMSA(PANASONIC)																							
WP9440	2	400	700	1000	10-60k	.06	.06	.1	.006	20-20k .5							5.69 19 17	75	\$2090.00		Glass epoxy main circuit boards, fan-blown heat sink tunnel, die-cast front panel for strength.		
WP9220	2	200	300		10-60K	.05	.05	.1	.006	20-20k .5 1						1.23	5.69 19 17	38.6	\$1090.00		Models 9220, 9110, 9055 feature instrumentation type electronically balanced/differential input circuits, five-way binding posts, phone jack outputs, overload/short circuit/DC and thermal protection, low feedback design, all Elna low ESR capacitors in power supply and audio path, all models feature a 5-year warranty.		
WP9110	2	100	150		10-85k	.05	.05	.1	.006	20-20k .5						1.23	3.94 19 15.94	28.6	\$840.00				
WP90S	2	50			10-85k	.05	.05	.1	.006	20-20k .5						1.23	2.3 19 15.69	19	\$590.00				

Model	Number of Channels		Cont. power/channel at 8 ohms all channels driven		Cont. power/channel at 4 ohms all channels driven		Cont. power/channel at 2 ohms all channels driven		Power Bandwidth Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, \$	Features
RANE CORPORATION																			
MA6	8	100	150		20-20k	.1	.1		.2	5-80k 3	1.23	5.25 19 11.5	44	\$1349.00	Built-in 15dB limiter, auto bridging for 300W per pair of channels, 2-speed thermal fan.				
RENKUS-HEINZ INC.																			
P1500	2	300	500	750		.1	.1		20-20k .5	1.55	3.5 19 13.75	40	\$1598.00	Circuitry to protect speakers against damage from hard clipping and high frequency oscillations.					
SOUNDCRAFTSMEN See our ad on page 23																			
PM860	2	210	315	450	20-20k	.05	.05	.008	.05	20-20k .1	1.2	5 8.5 14	20	\$599.00	High current design to allow stability with 2 ohm speaker loads.				
200X2	2	145	210	230	20-20k	.05	.05	.008	.05	20-20k .1	1	5.25 19 10.5	25	\$699.00	Level controls, bridging switch, balanced/unbalanced inputs. With LED power meters price is \$799.00.				
450X2	2	210	315	450	20-20k	.05	.05	.008	.05	20-20k .1	1.2	5.25 19 11.75	28	\$799.00	Same as above with LED power indicators.				
900X2	2	375	875	900	20-20k	.05	.05	.008	.05	20-20k .1	1.22	5.25 19 16.5	59	\$1599.00					
RA7501	2	275	420	320	20-20k	.05	.05	.05	.05	20-20k .1	1.21	7 19 15	47	\$949.00	Class H signal tracking design for maximum efficiency. RA7502 has LED power indicators for \$1049.00.				
SPECTRA SONICS																			
701	1- ∞	33	58	86	20-20k	.05	.075	.025	.025	DC-20k	1.37	2.5 10 1.875	.881	\$108.00	A modular amp for bi, tri, quad and multi-way amplification, used broadcast, recording.				
701BP	1- ∞	122	172	200	20-20k	.05	.075	.025	.025	DC-20k	1.37	5 10 1.875	1.76	\$216.00	Two 701s bridged together with the same qualifications.				
712B	2	30	50	80	20-20k	.05	.075	.025	.025	DC-20k	.707	5.5 19 14.5	22	\$595.00	A 3.5-inch rack mount, self contained power amplifier.				
712	2	100	100	100	20-20k	.05	.075	.025	.025	DC-20k	.707	5.5 19 14.5	24	\$760.00	Same physical features as above.				
STUDER REVOX See our ad on Cover II																			
A68	2	100	175	30-				0.1	30-20k 1	5.25 19 19.75	48.4	\$2000.00	Low closed-loop feedback eliminates Can operate in mono for 400W at 8 ohms.						
SYMETRIX, INC.																			
A-220	2	20	20	40	20-20k		.01	.05	20-20k 1	.5	1.75 19 8	9	\$315.00	Professional headphone and near-field monitoring amp, XLR balanced and unbalanced inputs.					
TANNOY NORTH AMERICA INC.																			
SR140	2	140	225		20-20k		.03	.01	20-20k .1	1	5.25 19 12	26	\$995.00	High definition, high current MOS-FET, convection cooling, designed for all reference monitoring.					
SR740	2	425	750	950	20-20k		.05	.04	20-20k .1	1.22	5.25 19 16.5	60	\$1995.00	Built-in processor which monitors and protects amps and speakers, dual speed forced air cooling.					
SR840	2	250	450	650	20-20k		.03	.01	20-20k .1	1.1	5.25 19 19.25	53	\$3498.00	High definition, high current MOS-FET amplifier designed for mastering and mixing, convection cooled.					

Model	Number of Channels		Cont. power/channel at 8 ohms, all channels driven		Cont. power/channel at 4 ohms, all channels driven		Cont. power/channel at 2 ohms, all channels driven		Power Bandwidth	Hz-kHz	IM at 1 watt, %	IM at full power, %	THD at 1 watt, %	THD at full power, %	Frequency Response at 1W +/- dB	Sensitivity for full output, V	Dimensions, H/W/D, in.	Weight, lbs.	Price, \$	Features
3rd GENERATION																				
FL1200	2	380	625	20-20k							.002	.007	10-55k	.8	3.5 19 16	26	\$3495.00			
HP1000	2	345	525	20-20k							.02		20-30k	.8	5.25 19 16	44	\$1859.00			
VL600	2	200	320	20-20k							.002	.007	10-55k	.8	3.5 19 16	21	\$2359.00			
HP400	2	162	225	20-20k							.02		20-30k	.8	5.25 19 16	34	\$1159.00			
SP200	2	84	105	20-30k							.02		20-30k	.8	5.25 19 11	32	\$899.00			
QL200	2	84	115	20-20k							.002	.007	30-45k	.8	1.75 19 16	20	\$1695.00			
YAMAHA MUSIC CORPORATION, USA See our ad on page 10																				
PD2500	2	250	360	500	20-50k						.007	.007	10-50k	1.23	3.88 18.88 18.88	26.5	\$1395.00			High power, 1000 watts in bridged operation.
PC2002/2002M	2	240			20-100k						.01	.003	10-50k	.7	7.25 18.88 16.25	44	\$1345.00 \$1500.00			High power, 700 watts in bridged mono operation. PC2002M has meters.
P2250C	2	170	250		10-50k						.01	.005	10-50k	1.23	5.25 18.88 16.75	41.8	\$795.00			XLR and .25-inch inputs, barrier strip with prov. for input X-formers, forced air cooling.
P2150C	2	100	150		10-50k						.005	.005	10-50k	1.23	5.25 18.88 16.75	37.4	\$625.00			XLR and .25-inch inputs, barrier outputs with prov. for input X-formers, forced air cooling.
P1250C	1	170	250		10-100k						.005	.005	10-50k	1.25	5.25 18.88 16.75	33	\$550.00			XLR and .25-inch jacks, barrier strip outs, P250T output X-former may be installed internally.
P2075C	2	50	75		10-50k						.05	.005	10-50k	1.23	3.88 18.88 14.38	21	\$625.00			XLR and .25-in. inputs, barrier strip output with provision for input transformers.
P1150C	1	100	150		10-100k						.005	.05	10-50k	1.23	5.25 18.88 16.75	28.6	\$550.00			XLR and .25-in. ins., barrier outs, P150T output X-former may be installed internally, forced air.
PC1002	2	100	150		10-110k						.01	.01	10-50k	.7	5.5 18.88 13.25	34.2	\$895.00			XLR and .25-inch input jacks, high power-300 watts in mono.
YORKVILLE SOUND, INC.																				
Beta 150	1	90	150		20-20k							.04	20-20k .5	1	6 21.5 9	22	\$315.00			Rack mountable, in wooden case.
Beta 150EQ	1	90	150		20-20k							.04	20-20k .5	1	6 21.5 9	22	\$375.00			10-band graphic equalizer, rack mountable, in wooden case.
Beta 500	1	170	300	500	20-20k							.05	20-20k .5	1.5	7.5 21.5 9	40	\$585.00			In wooden case.
Beta 800	2	200	400		7-30k							.05	7-30k 1	1.4	7.5 21.5 15	60	\$995.00			Rack mountable, in wooden case.
AP500	2	150	250		20-22k							.1	20-22k .5	1.2	3.5 19 13.5	39	\$775.00			Balanced inputs, binding posts and .25-inch (phone) outputs, rack-mountable.

ADDRESSES

Altec Lansing Corporation
10500 W. Reno
Oklahoma City, OK 73128

Audio Media Research (AMR)
PO Box 1230
Meridian, MS 39301

Biamp Systems
14270 NW Science Park
Portland, OR 97229

Boulder Amplifiers
Silver Lake Research
4850 Sterling Dr
Boulder, CO 80301

Carver Corporation
PO Box 1237
Lynwood, WA 98046

Carvin Corporation
1155 Industrial Ave
Escondido, CA 92025

Crest Audio
150 Florence Ave
Hawthorne, NJ 07506

Crown International
1718 W. Mishawaka Rd
Elkhart, IN 46517

dbx
71 Chapel St
Newton, MA 02195

Electro-Voice Inc.
600 Cecil St
Buchanan, MI 49107

David Hafler Company
5910 Crescent Blvd
Pennsauken, NJ 08109

Hill Audio
5002 N. Royal Atlanta Dr #B
Tucker, GA 30084

HM Electronics
6675 Mesa Ridge Rd
San Diego, CA 92121

Industrial Research Products
321 Bond St
Elk Grove Village, IL 60007

JBL Professional
8500 Balboa Blvd
Northridge, CA 91329

Peavey Electronics Corporation
711 A St
Meridian, MS 39301

QSC Audio Products Inc.
1926 Placentia Ave
Costa Mesa, CA 92627

Ramsa-Panasonic
6550 Katella Ave
Cypress, CA 90630

Rane Corporation
10802 47th Ave W.
Everett, WA 98204-3400

Renkus-Heinz Inc
17191 Armstrong Ave
Irvine, CA 92714

Soundcraftsmen
2200 S. Ritchey
Santa Ana, CA 92705

Spectra Sonics
3750 Airport Rd
Ogden, UT 84405

Studer Revox
1425 Elm Hill Pike
Nashville, TN 37210

Symetrix Inc
4211 24th Ave W.
Seattle, WA 98199

Tannoy North America Inc
300 Gage Ave Unit #1
Kitchener, Ontario
N2M 2C8 Canada

3rd Generation
431 Hwy 165
Voluntown, CT 06384

Yamaha Music Corporation USA
PO Box 6600
Buena Park, CA 90622

Yorkville Sound Inc
56 Harvester Ave
Batavia, NY 14020

Recording Techniques

BRUCE BARTLETT

RECORDER/MIXER FEATURES

● Remember how complicated and expensive it used to be to record a demo tape? It required a studio full of fancy equipment, a skilled recording engineer, and many long hours learning the technology. Nowadays, you can put your musical ideas on tape quickly, easily, and with little expense—thanks to cassette recorder/mixers.

A multi-track cassette recorder/mixer (portable studio or mini-studio) is a combination 4-track cassette recorder and mixer in one small, portable package. The cassette unit records up to four individual tracks, each track containing the signal of a different musical instrument, or different combinations of instruments. After recording all the instruments and vocals on these tracks, you mix them down to two-channel stereo.

Despite the name, “portable studio,” such a device is not complete in itself. You also need microphones, mic stands, speakers or headphones, and possibly a delay unit, reverb unit, and compressor for the vocals.

Also, you must set up a stereo mix whenever you want to hear what’s on tape. Rather than doing that, you can record the mix onto an external cassette deck or open-reel recorder.

While recorder/mixers offer good performance for their price, they really can’t compete in sound with any decent demo studio. Cassette recorders have less headroom, more noise, and narrower frequency response than open-reel recorders. Still, recorder/mixers are yours to use whenever the mood strikes; they’re affordable, and you can communicate your music with them.

RECENT PRODUCTS

Recorder/mixers currently on the market are made by Fostex, Tascam,

Audio Technica, AMR, Yamaha, Cutek, Vesta Fire, Ross, and Clarion.

There are at least three levels of recorder/mixers:

An input module is a group of controls in the mixer that affects a single input signal.

1. The personal recorder/mixer (about \$449) has two inputs and records up to two tracks at a time, but has no control of special effects. Features and sound quality are limited, but are adequate for recording musical ideas for your own amusement and practice.

2. The budget recorder/mixer (about \$449 to \$1095) has four inputs and records up to two or four tracks at a time. Some units have an effects loop so that you can add special effects such as digital reverb. Most but not all have equalization or tone control. Sound quality is good enough for an audition tape to get gigs, but probably not quite good enough for a demo tape to send to a record company.

3. The more-advanced recorder/mixer (\$1300 and up) records up to four tracks at a time, has four-to-eight inputs, has equalization, and allows control of special effects. It is good enough to make demo recordings of yourself or other musicians.

OVERVIEW OF RECORDING/OVERDUBBING/MIXDOWN

Before describing recorder/mixer features, we need to know the three stages in making a multi-track recording:

● Recording: You record one or more instruments onto one or more tracks. Usually the rhythm instruments—drums, bass, guitar—are recorded first.

● Overdubbing: While listening to pre-recorded tracks over headphones, musicians add new parts that are recorded on open (unused) tracks. Vocals and quiet acoustic instruments are usually overdubbed.

● Mixdown: Once all the tracks are recorded, you mix or combine them into 2-track stereo. This stereo mix is recorded onto an external 2-track cassette or open-reel deck. This recording is the final master tape, which can be duplicated on cassette or record.

RECORDER/MIXER MIXER SECTION

Let’s cover the features of the recorder/mixer in detail.

The mixer portion has several functions. It:

1. Amplifies the signals from all the microphones,
2. Mixes them in various combinations,
3. Routes them to tape tracks,
4. Adjusts the tone quality and stereo position of the instruments, and
5. Controls special sound effects.

The mixer can be divided into three sections:

1. Input modules
2. Output module
3. Monitor section

INPUT-MODULE FEATURES

An input module (*Figure 1*) is a group of controls in the mixer that affects a single input signal. The module is usually a narrow vertical strip, one per input. In cassette recorder/mixers, the number of input modules ranges from 2 to 8, with 4

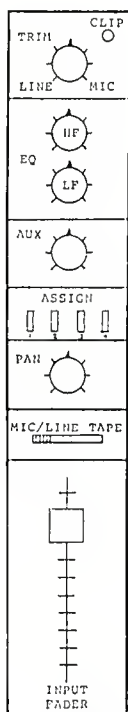


Figure 1. A typical input module.

being the most common. The more inputs, the more mics you can use during a recording. If you're recording only yourself, you may need only two inputs.

Figure 2 is a block diagram showing the signal flow from input to output through a typical input module, and through the other circuits in a mixer. Here's a description of each component of the diagram, going mostly from left to right (input to output):

- **Input connectors.** These connectors are for microphones and other signal sources. In some recorder/mixers, a single 1/4-inch phone jack is used both for mic-level and line-level signals. A mic-level signal (from a microphone) is about 1-to-2 millivolts nominal. A line-level signal (from a synthesizer, drum machine, or direct from an electric guitar) is about 0.3 to 1.23 volts.

If a single jack is provided for both mic and line signals, the two levels are handled either by a MIC/LINE switch, a HI-LO GAIN switch, or a TRIM control that reduces the line-level signal to prevent distortion.

Some units have separate jacks for mic and line inputs. The mic inputs are either unbalanced 1/4-inch phone jacks or XLR-type. The phone jack costs less but is more susceptible to

pickup of buzz and hum. This is not a serious problem in most small studios. The line jack is either a 1/4-inch phone jack or an RCA phono jack. You can plug an electric instrument directly into such an input without using a direct box, if the cable is short to prevent hum.

Some newer recorder/mixers have a SYNC input. It goes into track 4, and is used for recording a special tape-sync signal from a computer running a sequencer program. It's becoming common, in the final mix, to combine sequencer recordings of MIDI-equipped synthesizers with tape recordings of real instruments and vocals. The tape-sync feature lets you synchronize tracks recorded on tape with sequencer tracks recorded in computer memory. It also lets you overdub two or more sequencer tracks onto tape by keeping them synchronized.

- **Input selector.** This switch selects the input you want to process: either mic, line, or tape. The switch might be labeled in one of these ways:

Mic/line/tape
Mic/line/remix
Mic-line/off/tape
Input/mute/track
Input/off/line

The various switch positions work as follows:

Mic: The mic signal enters the mixer.

Line: The line signal enters the mixer.

Mic-line or Input: Either the mic signal or the line signal enters the mixer, depending on what is plugged into that input.

Tape, Track, or Remix: The tape-track signal enters the mixer (for overdubbing or mixdown).

Off or Mute: No signal is processed. During mixdown, it's a good idea to mute tracks that have nothing playing at the moment to reduce tape hiss.

- **Microphone preamp.** This amplifier inside the input module boosts the weak microphone signal up to a stronger line-level signal.

- **TRIM (GAIN).** Trim is an extra volume control that adjusts for a wide range of input signal levels. There are two ways to set the TRIM control: by observing the LED overload indicators, or by observing the meters. LED overload (clip or peak) indicators are little lights that flash when the input

signal level is so high that it causes distortion in the mic preamp. This occurs when a microphone is picking up a very loud instrument. If the overload light flashes, gradually turn down the TRIM control just until the light stays off.

Another way to set the TRIM control is as follows: Set the MASTER fader(s) to the shaded portion of its travel (sometimes at 0, about 3/4 up). Do the same for the INPUT fader. Assign the signal to a tape track, and adjust the TRIM control so that the meter peaks around 0.

Inexpensive recorder/mixers do not have overload LEDs or a TRIM control. The input fader serves this function.

Equalizers cover two or three bands of the audible spectrum.

- **INPUT fader.** After the microphone signal is amplified by the preamp, it goes to the input fader. This is a sliding volume control for each input signal. You use it during recording to set recording levels on tape, and during mixdown to set the relative loudness balance among instruments. In the Fostex X-15 II, the input faders are used during recording to set recording levels, and during mixdown as master volume controls.

- **Direct out.** Only the more elaborate units have this feature. This is an output connector following each input fader. The fader controls the level at the direct-output jack. It's used for recording one mic per track of an external multi-track recorder. Since direct out bypasses the mixing circuits farther down the chain, the result is a cleaner signal.

- **EQ (equalization).** The signal from the input fader goes to an equalizer, which means "tone control." With EQ you can make an instrument sound more or less bassy, more or less trebly, by boosting or cutting certain frequencies.

Equalizers cover two or three bands of the audible spectrum. Inexpensive units have a simple two-knob bass and treble control; you can boost or cut the treble or bass. Fancier models have a sweep-able EQ (sometimes called parametric EQ) that lets you "tune in" the exact frequency range

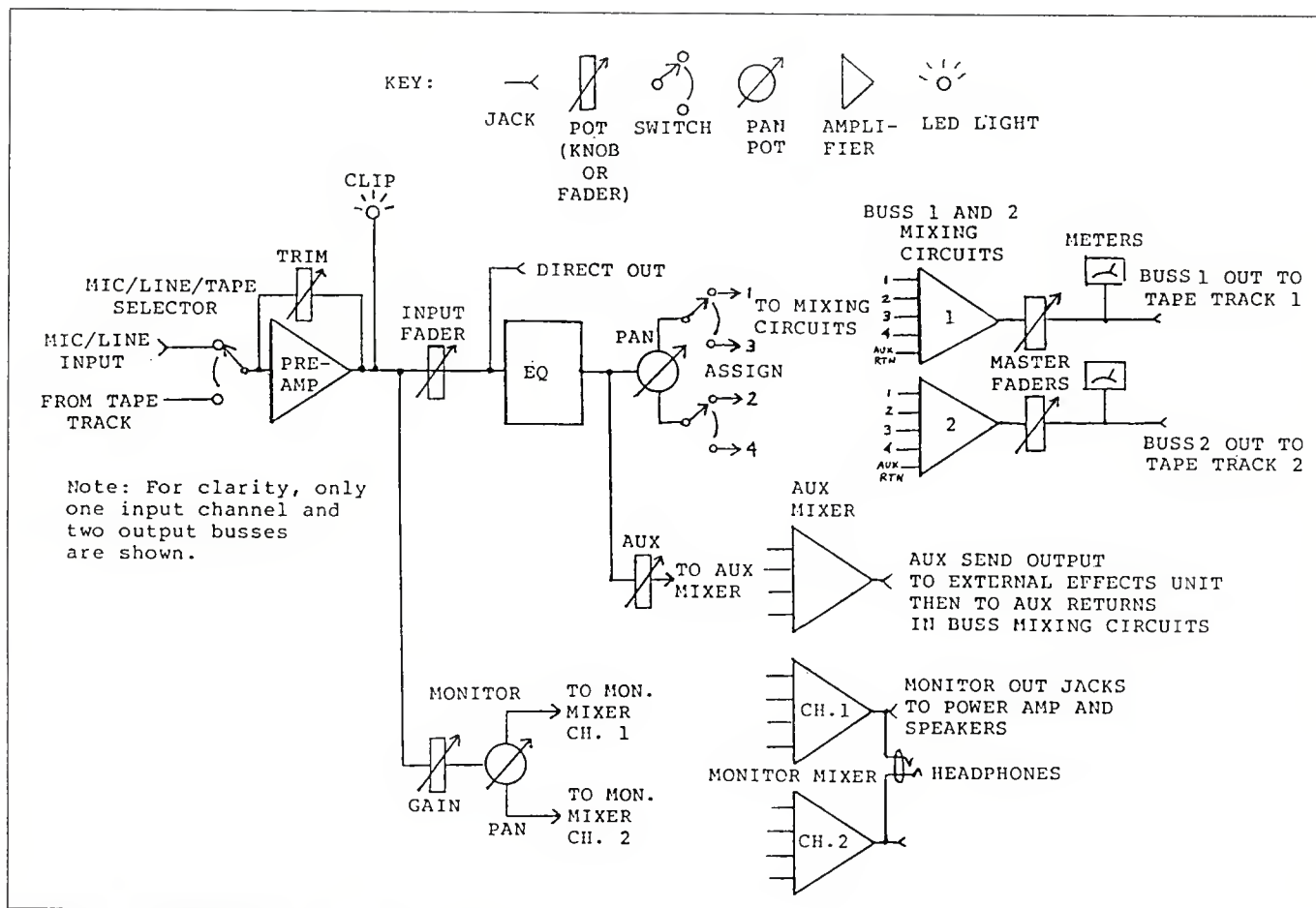


Figure 2. Signal flow in a typical mixer section of a recorder/mixer.

to work on. This feature adds cost and complexity, but gives you more control over the tone quality.

In inexpensive units, the EQ works on two inputs at a time during recording, and on the stereo mix during mixdown. This is less flexible than a unit with EQ on each input.

• **ASSIGN switches.** The equalized signal goes to a set of switches or knobs called the ASSIGN switches. These let you send the signal of each instrument to the desired tape track you want to record that instrument on. Some units have a track selector switch labeled "1, 2, 3, 4." Others assign tracks by using a combination of the PAN POT setting (explained next) and the record-select switches. Some inexpensive units always assign input 1 to track 1, input 2 to track 2, and so on.

• **PAN pot.** During recording and overdubbing, this is often used to assign inputs to tracks: left for odd tracks, right for even tracks. For example, if you set the pan pot left, you can record on tracks 1, 3, or both, depending on which record-select button you press. During mixdown, the

pan pot places the stereo image of each recorded track wherever desired between a stereo pair of loudspeakers. With the pan pot, you can locate an instrument at the left speaker, right speaker, or anywhere in between.

• **AUX (EFFECTS or FX).** Find this control in Figure 2 just after the EQ. During mixdown, this knob controls the level of an input signal sent to an external effects device, such as a digital delay or reverberation unit. The processed signal returns to the mixer, where it blends with the original signal, adding a sense of ambience or spaciousness to an otherwise "dry" track. This feature is essential if you want to produce a commercial sound.

During recording and overdubbing, the AUX knobs can also be used to create a mix heard over headphones. In this case, you don't connect the AUX SEND output to an effects unit. Instead, you connect it to a small amplifier that drives headphones. The headphone mix done with the AUX knobs is independent of the levels going on tape.

Some units have no AUX feature; some have one AUX send control; some have two. The more AUX sends you have, the more you can play with effects, but the greater the cost and complexity.

A few units have an AUX RECEIVE or AUX RETURN control for setting the level returning to the mixer. The processed effects signal enters the mixer through either a pair of AUX RETURN jacks or BUS IN jacks.

There may be a PRE/POST switch next to the AUX send knob. The PRE setting is used for a headphone mix during recording or overdubbing. The POST setting is used for effects during mixdown.

Some high-end recorder/mixers have PAN as well as GAIN for the AUX send.

The following features are omitted from Figure 2 for clarity.

• **OUTPUT fader or potentiometer.** In some inexpensive recorder/mixers, this control is found in each input module. It is used as a volume control

for that input or track in the monitor mix or headphone mix.

● **ACCESS jacks (INSERT jacks).** These let you plug a compressor in series with an input module's signal for automatic volume control. Inexpensive units omit this feature. Some units have access jacks on only two inputs.

The access jacks also can be used to insert a digital delay or reverb unit into the signal path of one track. On the delay/reverb unit, you set the dry/delay mix as desired by using its built-in MIX control. Normally, however, delay/reverb units are patched between the AUX send and receive jacks, with the MIX control on the delay/reverb unit set all the way to "delay" or "wet."

● **BOUNCE (PING-PONG or TRANSFER).** The bounce feature allows you to mix several pre-recorded tracks with your mixer and record the result on an empty track. Then the original tracks are erased, freeing them up for recording more instruments. For example, let's say you've recorded instruments on tracks 1, 2, and 3. You combine these tracks with your mixer and record the mix on track 4. Then you can record three more parts on tracks 1, 2, and 3.

● **RECORD/PLAY/SEND switch.** Found in some inexpensive units, this switch works as follows:

Record: Record this input signal on the same numbered track.

Play: Play this track (or make it safe—not able to be recorded or erased).

Send: Send this track to all the other input modules for bouncing. Only the track with "Record" pressed will record the bounced tracks. For example, if you want to bounce (transfer) tracks 1, 2, and 3 to track 4, press Send for tracks 1, 2, and 3; press Record for track 4.

OUTPUT-MODULE FEATURES

The output module includes mixing circuits, master faders, and meters. In the mixer, the output module is the final section that feeds signals to tape tracks. Let's look at each part in detail.

● **Mixing circuits.** Each channel or bus that feeds a tape track has one of these circuits. The bus-1 mixing circuit accepts the signals from all the inputs assigned to bus 1 and mixes them together to feed track 1 of the

tape recorder. The bus-2 mixing circuit mixes all the bus-2 assignments, and so on. Mixing circuits also accept AUX RETURN signals, such as the reverberated signal from an external digital reverb unit.

A four-bus mixer provides four independent output channels or buses; each bus carries a signal which may contain the sounds from one or more musical instruments. The four buses feed a 4-track cassette recorder. A mixer with only two output buses can be used with a 4-track recorder by recording two tracks at a time.

● **Master faders.** The output module also contains the MASTER faders (OUTPUT faders). These are one, two, or four faders that control the overall level of the output channels.

● **Meters.** Meters measure the strength (level or voltage) of various signals. Usually, each output bus has a meter to measure its signal level. Since these buses feed the tape tracks, you use the meters to set the recording level for each track. Your recorder/mixer will have one of these three types of meters:

A VU meter: A voltmeter that shows approximately the relative loudness of various audio signals.

An LED bargraph level indicator: A column of lights (LEDs) that show peak recording level.

An LED peak indicator: A light mounted in a VU meter. It flashes when peak recording levels are excessive.

The VU meter does not respond fast enough to musical attacks or peaks to indicate them accurately. The LED peak-reading meter is a more-accurate indicator of true recording level.

● **TAPE OUT jacks.** Some units have TAPE OUT jacks (not shown). These are connected to the tape-track outputs, and are used for copying your four-track cassette recordings onto multi-track studio recorders for further overdubs and processing.

MONITOR-SECTION FEATURES

The monitor section controls what you're listening to. It lets you select what you want to hear, and lets you create a mix over headphones or speakers to approximate the final product. This monitor mix has no effect on the levels going on tape.

● **Monitor mixer.** The monitor mixer (labeled MONMIX or TAPE CUE on some mixers) is a small submixer built into the larger mixer. It controls the balance among vocals and instruments heard over headphones or loudspeakers as you're recording.

The monitor mixer is made of several MONITOR GAIN and MONITOR PAN controls, plug two mixing circuits that feed a headphones or an external stereo amplifier and speakers.

As shown in *Figure 2*, the monitor gain (volume) and pan for each input signal come before the input fader (pre-fader). You use the monitor gain knob to control how loud each live instrument or track is in the monitor mix. You use the monitor pan knob to control the position of the monitored instrument or track between your stereo speakers. Some inexpensive recorder/mixers use the OUTPUT controls on the input modules to create the monitor mix.

The monitor mixer also blends pre-recorded tape tracks and live microphone signals into a "cue mix" that is sent to the musicians' headphones in the studio. The musicians record new parts while listening to the cue mix over headphones.

Note that the auxiliary (AUX) sends in the mixer can serve double duty as controls for a monitor mix or headphone mix. In most recorder/mixers, the monitor mix and cue mix are identical.

● **Monitor select.** The MONITOR SELECT buttons let you choose what signal you want to monitor or listen to. Since the configuration of these buttons varies widely among different models of recorder/mixers, they are not shown in *Figure 5-2*. Here's a listing of some of the MONITOR SELECT buttons you may find:

Monitor Tr. 1, 2, 3, 4. You select which track or combination of tracks you want to hear, and mix them with the output controls in the input modules.

Tape/bus 1, 2, 3, 4. You select whether you want to hear signals off the tape, or from the bus (the mixer output), for channels 1, 2, 3, or 4. Select "tape" to hear a playback, or for hearing previously recorded tape tracks during an overdub. Select "bus" to hear the live signal that you're recording.

Tape-bus/stereo/aux, or Tape-bus/2-tr/aux. The "tape-bus" switch position is described above. The "stereo" or "2-tr" switch position lets you hear the 2-channel stereo mix during mixdown. The "aux" position is to hear the aux signal (effects or headphone mix).

Remix/cue/aux. "Remix" is the 2-channel stereo mix you want to hear during mixdown. "Cue" is the headphone mix. "Aux" is the effects-signal mix.

Line/mixdown. This combination is found only in the Clarion unit. "Line" is for overdubbing. It lets you hear a mono mix of live signals and tape signals over headphones for selected tracks. "Mixdown" is used during mixdown to hear a stereo mix of all four tracks.

Some units have no monitor-select switches. Instead, you always monitor the 2-channel stereo monitor mix.

● *Headphone volume control.* This controls the loudness of the headphones. All but the least expensive units have this feature.

RECORDER/MIXER RECORDER SECTION

The 4-track cassette deck built into the recorder/mixer also has many features to investigate:

Chances are you'll be equally satisfied with Dolby C or dbx.

● *Overdubbing.* This is recording a new track in sync with old tracks. When a musician overdubs, he or she listens with headphones to previously recorded tracks, plays along with them, and records a new musical part on a blank track. In this way, instruments can be recorded one at a time until all the parts are on tape. This provides maximum control and best sound quality. You can be a one-man band by overdubbing all the parts yourself. Overdubbing is a standard feature in all multi-track recorders.

● *Synchronous recording.* This feature is used during

overdubbing to keep pre-recorded tracks synchronized with new parts being added "live." Pre-recorded tracks are played back from the record head, rather than the playback head. This keeps the timing of the old

and new musical parts in sync. Since all recorder/mixers combine the record and playback head into one, there are no sync problems.

● *Punch in/out.* With this feature, you can fix a mistake on a track without doing the whole track over. You insert corrected musical parts into a previously recorded track. As the track is playing, you "punch in" the record button at the appropriate spot in the tune. Then the musician plays a corrected version that is recorded over the previous performance on tape. When the musician has finished playing the corrected part, you "punch out" of record mode so the rest of the track is not erased. All recorder/mixers accept a footswitch so the musician can punch in and out while performing.

● *Noise reduction.* This is a circuit that reduces tape hiss. The Dolby and dbx systems are commonly used. Noise reduction is essential with cassette recorders because the slow tape speed and narrow track width result in audible tape noise. Noise reduction circuits clean up the signal.

Dolby C is more effective than Dolby B, and dbx is more effective than either. Still, Dolby is free of the "breathing" sound (modulation noise) that is sometimes heard on dbx encoded tracks of bass or bass-drum. Chances are you'll be equally satisfied with Dolby C or dbx.

● *Tape counter.* This counter is analog in low-cost machines and digital in higher-cost machines. The digital counter is slightly more accurate.

● *Return-to-zero, zero stop, autolocate, or memory rewind.*

With this function, the recorder automatically rewinds to a preset point marked "000" on the tape counter. It's useful for repeated practices of punch-ins and mixes.

Some high-end units can repeatedly shuttle back and forth between two pre-set points, say, at the beginning and end of an overdubbed section.

● *Tape-speed options.* A cassette recorder that operates at 1-7/8 in./sec. is compatible with commercial pre-recorded cassettes, so you can play them on your recorder/mixer. A 3-3/4 in./sec. recorder will not play standard pre-recorded cassettes correctly, and uses tape twice as fast, but it provides better sound quality (extended high-frequency response, less tape hiss, and less wow & flutter). Some

recorders offer selectable tape speeds.

● *Pitch control.* This varies the speed of the cassette recorder. This function lets you adjust the pitch of previously recorded tracks to match the tuning of new instruments to be added. Pitch variation ranges from ± 10 percent to ± 15 percent among different models.

TAPE-RECORDER SPECIFICATIONS

Wow & Flutter:

Wow is a slow periodic variation in tape speed; flutter is a rapid variation. If excessive, they wobble the pitch of recorded instruments. The lower the wow & flutter spec, the steadier is the reproduced pitch.

A wow & flutter specification of 0.03 percent RMS weighted (or WRMS) is excellent. 0.04 percent RMS weighted (or WRMS) is very good. 0.1 percent IEC/ANSI peak weighted is very good. Higher values than the above are not as good.

Signal-to-noise ratio:

This is the ratio, expressed in dB, between the recorded signal level and the noise level. Specifically, it is the ratio between a 400-Hz signal recorded on tape at 3 percent third-harmonic distortion, and the residual noise floor of the tape recorded without an input signal. The higher the figure, the more noise-free is the recording. All the following specs are measured with noise reduction, A-weighted:

90 dB (typical of dbx) is excellent.

70 dB (typical of Dolby C) is very good.

65 dB is good.

55 dB is fair.

Record/play response:

This is the range of frequencies that the recorder will record and play back at an equal level, within a tolerance (such as ± 3 dB). The wider the frequency response, the better the fidelity.

40Hz-12.5kHz ± 3 dB is fair (typical of personal recorder/mixers).

40Hz-14kHz ± 3 dB is good.

40Hz-18kHz ± 3 dB is excellent.

As we've seen, it doesn't cost much to get started in home multi-track recording. By understanding recorder/mixer features, you'll be able to find one well-suited to your needs—and your budget. □□

Hands On

TASCAM PORTA-05 MINISTUDIO

THE PORTA-05 IS TASCAM'S LATEST OFFERING FOR THE home-recording market: a low-cost, portable, 4-track cassette recorder/mixer. It's a convenient tool for making home demo tapes, or for recording musical ideas while composing.

Although the Porta-05 is very compact, it has a wide range of features. With it you can record up to four input signals at a time onto one or two tape tracks. Each input can be assigned to any track. You can overdub, punch in/out, bounce tracks, and mix the four tracks down to 2-track stereo with effects. There's also a SYNC feature: You can record an FSK tape-sync signal on track 4, and use it to synchronize sequencer recordings of MIDI keyboards and drum machines that are recorded on different tape tracks.

Other features include dbx noise reduction, headphone-level control, analog tape counter, return-to-zero, pitch control, punch-in footswitch jack, and 1-7/8 in./sec. tape speed. The Porta-05 retails for \$495.

CONNECTORS AND CONTROLS

Four input connectors are included. Each is a 1/4-inch phone jack for unbalanced microphone- or line-level signals. A synthesizer, electric guitar, drum machine, or other line-level signal can plug directly into one of these jacks for direct-injection recording. Inputs 1 and 2 can handle either mic or line signals, with the level difference handled by a TRIM control. Inputs 3 and 4 accept only line-level signals.

Each input module includes a fader, TRIM control (modules 1 and 2 only), PAN pot, EFF (effects) send control, and TAPE CUE control. The futuristic, attractive, controls operate smoothly. There is no source-selector switch. Instead, you hear the source when you plug it into the Porta-05; you hear the tape when the source is unplugged. Although this arrangement reduces cost, it's a

little confusing. If you forget to unplug your input cables during mixdown, you won't hear the tape tracks playing.

The input fader is used during recording to set record levels, and during mixdown also to adjust the balance among tracks. The TRIM pot adjusts the pre-amplifier gain to accommodate a wide range of input levels, from low-level microphone signals to high-level line signals.

During recording, the PAN pot is used along with the RECORD FUNCTION switch to assign each input to the desired track. If you rotate the PAN pot full left, the input signal goes to track 1 or 3. If you rotate it full right, the signal goes to track 2 or 4. It's hard to tell at a glance which track you're assigned to, so you'll have to be careful. During mixdown, the PAN pot is used to place the phantom image of each track wherever desired between the pair of playback

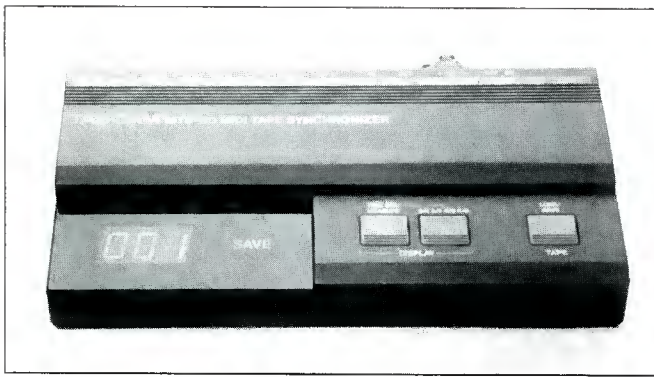


speakers—left, center, right, or anywhere between.

The EFF control adjusts the amount of signal going to an external effects device, such as a digital reverb or delay unit. It also could be used to set up a cue mix for the musicians' headphones.

The TAPE CUE control works only during playback or overdubbing. It sets the level of recorded tracks heard over the monitor speakers or headphones. Using the TAPE CUE controls, you can set up a mono cue mix of recorded tracks and live signals being recorded. The level of the live signals in the cue mix is determined by the fader settings. Since these faders are used to set recording levels, you mix the recorded tracks relative to the fixed live signals.

The output module includes a MASTER fader to control overall level, LED bargraph level indicators to show recording level, an EFF RTN (effects return) control to set the overall level of a mono signal returning from the external effects unit, and a 2-stage equalizer for bass and treble



The Tascam MTS-30 MIDI Tape Synchronizer

control. This equalizer works during recording on individual tape tracks, and during mixdown on the overall stereo mix. The effects return is mono, so you cannot have stereo reverberation—the reverb is centered between your playback speakers.

A MONITOR SELECT switch lets you monitor (listen to) either CUE (a mono mix of the four tracks), REMIX (the 2-track stereo mix), or EFF (the effects mix sent to the external effects unit). The METER switch allows metering of INPUT (the level of each of the four inputs) or BUS (the 2-channel output bus). Input metering is pre EQ; bus metering is post EQ. When you meter BUS, you also can meter the effects send level.

The metering system can be confusing until you get used to it. There are four LED bargraph meters, but these do not necessarily correspond to the four tape tracks. Here's an example: Suppose you record input 2 on track 3. If you meter INPUT, the number 2 LED meter will light, because you're putting signal into input 2. If you meter BUS, the number 2 LED meter will light again, because that meter shows the level of bus 2 (right) which feeds tracks 3 and 4.

Cassette-recorder controls include a dbx on/off switch, pitch control for up to ± 15 percent pitch variation, an analog tape counter, and a zero-return switch. If you turn on zero-return and hit REWIND, the cassette automatically rewinds to the point marked "000" on the tape counter and stops. This is useful for repeated practices of overdubs or mixdowns. A cassette is ejected manually by lifting the cassette cover. The Porta-05 is biased for chrome tape. Output connectors are all RCA phono jacks. There's a pair for the 2-track stereo bus labeled LINE OUT, an EFFECT OUT jack, an EFFECT RTN jack, and a SYNC OUT jack (the track-4 output). The SYNC output can be connected to a tape sync connector on a computer MIDI interface, or can be connected to the TAPE IN connector on the MTS-30 MIDI/FSK Translator.

The unit is powered by a plug-in 12V adapter. Putting the power supply outside the main unit reduces size, weight, and hum. Operating instructions are excellent: clear, thorough, and concise, with helpful diagrams.

PUBLISHED SPECIFICATIONS

These cassette-recorder specifications are followed by my evaluation:

Record/play frequency response: 50Hz-12.5kHz ± 3 dB. (Good but not excellent. There will be some loss of extreme high frequencies, which dulls cymbals and percussion slightly).

Signal-to-noise ratio: 60dB with dbx noise reduction. (Very good, but not excellent. Some slight hiss will be audible). Wow & Flutter: 0.05 percent weighted. (Good but not excellent. The reproduced pitch of instruments will waver very slightly, almost undetectably).

The Porta-05 performs very well for its price range.

OPERATION

The Porta-05 is easy to operate. I completed a multi-track recording, overdubs, and mixdown in half-an-hour. A novice should be able to master the controls in an hour.

Let's go over the procedures for each stage of a recording session:

Recording

1. Plug in headphones or a power amplifier and speakers.
2. Plug in microphones and/or electric instruments.
3. Turn on dbx noise reduction.
4. Turn up the PHONES level and monitor CUE.
5. Set the METER switch to BUS.
6. For each input in use, set the PAN pot and RECORD FUNCTION switch to assign the input to the desired track(s). Remember you can record on only two tracks at a time.
7. Set the MASTER fader and input faders to the shaded areas.
8. While the musician is playing, set the TRIM control to adjust the recording level. The LED meters should peak at 0 dB on the loudest parts. If you're recording on tracks 3 or 4, set the recording level with the input fader.
9. Set EQ (bass and treble) as desired. You may need to readjust the recording level after setting EQ.
10. Reset the tape counter to "000." Set the ZERO RETURN switch to ON.
11. Press the RECORD button to start recording.

Playback

1. Rewind the tape. It will stop automatically at the "000" point on the tape counter.
2. Press the PLAY button.
3. Adjust the TAPE CUE controls for the desired mix.

Overdubbing

1. Plug in the cable for the voice or instrument you're going to overdub.
2. Rewind the tape. It will stop automatically at the "000" point on the tape counter.
3. Monitor CUE; meter BUS.
4. Using the PAN pot and RECORD FUNCTION switch, assign the new instrument to be overdubbed to the desired open track.
5. Set the MASTER fader and input fader to the

shaded areas.

6. Have the musician play. Using the TRIM control, set the recording level to peak around 0 dB during the loudest part.

7. Press the PLAY button.

8. Using the TAPE CUE controls, set up a headphone mix of the recorded tape tracks and the live signal being recorded. The monitor level of the live signal is set by the input fader, which also is used to set the recording level.

9. Rewind the tape, either to the beginning, or just before the point where you want to start overdubbing.

10. Record the overdub while monitoring the cue mix.

11. Punch in/out as needed.

Bouncing tracks

1. Unplug all input cables.

2. Monitor REMIX; meter BUS.

3. Using the PAN pots and RECORD-FUNCTION switch, assign the tracks you want to bounce to an open (unused) track.

4. Play the tape.

5. Using the input faders, adjust the mix. Add effects if needed by using the EFF controls. If necessary, add EQ with the bass and treble controls. These controls affect the overall mix, not individual instruments.

6. When the mix is satisfactory, rewind the tape and hit RECORD.

7. When the song is done, your bounce is complete.

Mixdown (Remix)

1. Unplug all input cables.

2. Connect a 2-track recorder (open-reel or cassette) to the LINE OUT jacks.

3. If you want to monitor through loudspeakers, connect the line out jacks on your 2-track recorder to your power amplifier inputs; connect the power amp outputs to your monitor speakers.

4. Connect a reverb or delay unit (or other effects device) between the EFFECT OUT and RTN jacks.

5. Monitor REMIX; meter BUS.

6. Play the multi-track tape.

7. Using the input faders, set the mix and record levels. Both the Porta-05 and the 2-track external deck should peak around 0.

8. Adjust pan, effects, and EQ as desired. The effects-send level is shown on the EFF meter, and should peak around 0.

9. When you're satisfied with the mix, rewind the tape. Play the multi-track tape and record the mix onto the external 2-track recorder.

SYNC-TRACK RECORDING

The Porta-05 can record on track 4 an FSK (Frequency-Shift-Keyed) tone from a computer MIDI interface, or from the optional Tascam MTS-30 MIDI/FSK Translator. This tone can synchronize sequencer playbacks during overdubs. That is, it lets you overdub various sequences

onto tape and keeps them in sync with each other. The sync tone makes all the recorded sequences start playing at the same time, and makes them all play at the same tempo.

With the MTS-30 FSK/MIDI Translator (\$225 retail), you can shuttle the tape to any location within a song and have your MIDI keyboards, drum machine and sequencer sync-up when you start the tape recorder. In contrast, most sequencers and drum machines only play back in sync from the beginning of a composition, due to the lack of song-pointer data. The MTS-30 takes MIDI timing information generated by a MIDI drum machine, rhythm composer or sequencer and translates it into FSK sync tones that can be recorded on tape. During playback, the MTS-30 reads the FSK tones from the recorder and translates them back to MIDI clocks that the system uses to stay in tempo (sync) with the tape recorder. It even keeps track of which measure you're in.

The MTS-30 includes error-correction circuitry, a large LED measure-number display and an automatically switched MIDI OUT/THRU terminal. This terminal lets you drive a drum machine while recording the sync tone from your sequencer, and to drive both the drum machine and sequencer on playback without repatching.


To record (stripe) a sync tone, connect the TAPE-SYNC connector from a computer MIDI interface (or the TAPE OUT connector from the MTS-30) to the track-4 input. With dbx switched in, record a sync tone slightly longer than the longest sequence you plan to record. The recording level of the sync tone should be -10 to 0 dB. With some sequencers, the operation of the SYNC function is very level-dependent—you'll have to experiment to find an appropriate sync-tone recording level. If the level is too high, the tempo of the sequencer playback will be too fast. If the level is too low, you may lose sync.

I had an unusual synchronization problem when using a Syntech Studio-One sequencing program (now discontinued) with the Porta-05. When I recorded audio on track 3 after having recorded the sync tone on track 4, crosstalk between tracks 3 and 4 confused my Passport MIDI interface and messed up the synchronization. Dave Oren, Tascam's Director of Product Development, helpfully suggested that I record audio on track 3 and tone on track 4 simultaneously. That worked. According to Dave, such a problem has not occurred with other sequencer/interface combinations. An improved version of the Studio-One sequencing software is currently offered by Sonus.

I also was unable to achieve sync using the MTS-30 with my particular MIDI system because, Dave explained, my synth and sequencer do not implement song position pointer. The MTS-30 works fine with most other systems.

SOUND QUALITY

Overall, the sound quality of the Porta-05 Ministudio is very good for the price, with some reservations. Noise modulation is audible on some tracks as a fuzzy sound accompanying the music. Extreme high frequencies are missing, causing some sounds to lose delicacy and crispness. Flutter and noise are very slightly audible.

Still, the Porta-05 performs very well for its price range. While it may not sound good enough for a demo you might send to a recording company, it is quite adequate for a demo tape played for customers to display your band's talents. The Porta-05 is also a great tool for composers using MIDI keyboards, and takes up little space. You'll really appreciate its convenience and ease of use. 

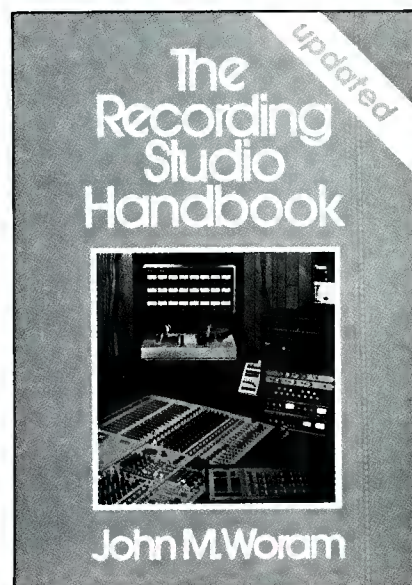
For Your Audio Library

The Recording Studio Handbook

by John Woram

\$39.50

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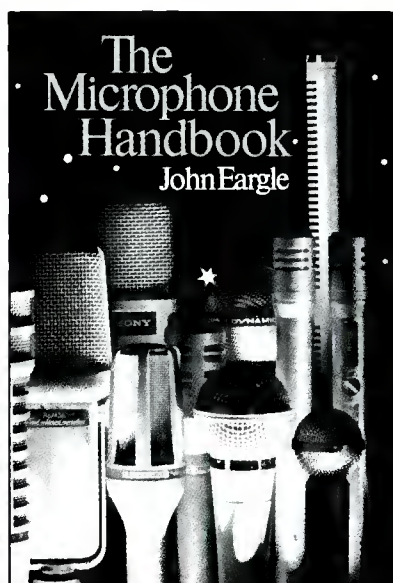


The Microphone Handbook

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● Sweeping changes in the tax law resulting from the Tax Reform Act of 1986 now make the year-end review of every recording, broadcast and sound contractor's tax situation more important than ever. Tax rates have changed this year as well as the rules governing the deductibility of common expense items such as interest and taxes.

At the root of any tax planning strategy is the accounting method that is employed by the recording, sound or broadcast operation. Although tax rates are reduced to only five last year and two in 1988 (not counting the 33 percent phase-out rate), a contractor's tax burden may be lighter if income can be spread equally throughout the years. Obviously, it is impossible for most sound engineers to arrange to receive income at a uniform rate, but much can be done to control taxable income by using the various methods of accounting that are available today.

For instance, a so-called "cash-basis" contractor reports income and deductions as they are received or paid. An "accrual-basis" studio reports these items as they become due. This year, the question of which method to use is taken from the hands of virtually every corporation, partnerships that have a corporation as a partner and tax shelters which are all now required to use the accrual method of accounting for tax purposes.

THE BEST ACCOUNTING METHOD

Sole proprietorships, Subchapter 'S' corporations and other partnerships are free to use the method best suited to their operations—and the one

PLANNED TAX SAVINGS

which will keep their tax bills low, year after year.

**the main difference
between a partnership
and a corporation is that
the latter is a taxable
entity separate and
distinct from its owners
and shareholders.**

In selecting the most suitable accounting method, one disadvantage of the accrual-basis should be considered—it is more difficult to shift items of income and expenses from one year to another. The cash-basis recording operation may be able to collect fees, rents, interest and other obligations in advance or in part. The cash-basis operation can also usually control expenses to some extent by accelerating or deferring payments for items such as advertising, supplies, repairs, interest and taxes.

**The profits are taxed to
the shareholders only if
and when they are
distributed to them in the
form of dividends.**

Control of this sort is not really as easy for the accrual-basis studio. These operations can, however, defer income by billing as little as possible during the closing days of a year in

order to reduce income for that year. Or, they can accelerate expenses by requesting the delivery and billing of supplies before the end of the year.

Regardless of the business form utilized by the business, major expenditures made or expected to be made before December 31st create a potential problem. It should be remembered that unlike the costs of running a business, which are tax deductible currently, expenditures for items of a more permanent nature (i.e., lasting more than a year) generally must be capitalized.

Usually the expenditure is either a currently deductible business expense or a capital expenditure that may be deductible over a period of time. Normally, the sound or recording contractor has no choice as to how he may treat it—only when the newly acquired property will be placed in service. Of course, contractors do have the option of currently expensing up to \$10,000 in costs for property that otherwise would have to be capitalized and depreciated, but that is a decision that does not have to be made until the income tax returns are prepared.

The end of the tax year is also an excellent time to take a look at the recording or sound business entity itself. For instance, all tax items generated by a business or profession operated by a sole proprietor are taxed directly to him or her.

From a tax standpoint, the main difference between a partnership and a corporation is that the latter is a taxable entity separate and distinct from its owners and shareholders. This is not true in the case of a partnership—a partnership does not pay tax. Rather, the partnership merely reports its income, the distributive

shares of which are attributed to the partners, the same as though they had been received without the intervention of the partnership.

CORPORATE DISADVANTAGES

A corporation has a distinct tax disadvantage in that its earnings are ordinarily taxed twice—once to the corporation when earned and then again to the shareholders when received in the form of dividends. Income earned by an ‘S’ corporation (which is taxed in much the same manner as a partnership) is taxed directly to the shareholders.

In comparing the tax factors in operating a sound business as a partnership or a proprietorship, rather than as a corporation, it should be remembered that not all of the corporate income will be subject to double taxation. The operators of a recording business may “withdraw” reasonable salaries from the corporation or have the recording corporation “repay” loans made to the entity by the shareholders.

A word of caution, however: Profitable sound corporations that have not paid dividends may not be able to deduct the full amount of salary payments to officer-stockholders. The failure to pay such dividends can cause a portion of the compensation of employee-stockholders to be treated as non-deductible dividends, even though the total payments considered as compensation are reasonable in amount.

On a similar note, because a corporation is a taxable entity separate and distinct from its shareholders, its profits are not, as is usually the case with unincorporated sound businesses, taxed to the owners when they are earned, although they are taxed to the corporation at that time. The profits are taxed to the shareholders only if and when they are distributed to them in the form of dividends. Therefore, to a limited extent, shareholders of a corporation have an advantage over partners or proprietors in that they may distribute the profits in the year or years in which the profits will be subject to the lowest individual tax liability. If the accumulations are unreasonable, however, a unique accumulated earnings tax will apply in addition to the regular corporate tax.

Should the recording business be operating as a sole proprietorship, the self-employed owner may employ a spouse, setting the stage for an employee exemption from Social Security taxes. Most sole proprietors are familiar with the general rule that services performed by an individual in the employ of his or her spouse are not considered “employment” for Social Security (FICA) or unemployment (FUTA) tax purposes. Consequently, the self-employed sound business person who hires his or her spouse need not pay or withhold Social Security taxes on spouse/employee wages.

**It would be pointless to
defer income from 1987
to 1988 if the 1988
income will be subject to
a higher rate of tax.**

Remember, however, the employee/spouse who is not covered under the Social Security Act will not be earning credits for the normal Social Security benefits as do other compensated employees.

RESPONSIBILITY FOR EMPLOYEES

Still on the subject of employees—regular, not spousal employees—employers today may be faced with the need to make substantial expenditures relating to the health and safety of their employees. Even though such an expenditure may be required under law, it is not currently tax deductible if it is otherwise a capital expenditure.

**if the tax is ignored rather
than legally postponed,
the studio owner will have
not a loan but a debt to
society that must be paid.**

Thus, whether or not a particular expenditure for health and safety is currently deductible depends on the nature of the expenditure. If it is a repair, it is currently deductible; if it

is a capital expenditure, it must usually be recovered through depreciation.


Splitting sound or recording business income between related entities and individuals is a common method of reducing the related group's aggregate tax liability. The most common device used to accomplish this income spreading is the payment of salaries to officer-shareholders of the studio corporation, thereby reducing taxable income.

In addition, there are other methods of splitting income among related entities. These include the financing of corporate operations by means of interest-bearing loans by stockholders, the leasing of business assets from stockholders and even the splitting of a studio corporation into several entities.

Tax planning primarily concerns the timing and the method by which the contractor's income is reported and the tax deductions and credits claimed. Our tax law permits every recording or sound business owner to select among various options in the reporting of income and the claiming of deductions and credits. The studio's task is to decide which of those options will minimize his or her tax bill.

In order to make this decision intelligently, the studio owner must have a fairly accurate picture of the operation's tax situation not only for the current year but also for the next year. It would be pointless to defer income from 1987 to 1988 if the 1988 income will be subject to a higher rate of tax.

Thus, the overall goal of any tax planning is to time taxable income so that it will fall in years when it will be subject to the lowest tax and to time deductible expenses to fall in years when it will offset income subject to a high tax rate. If the owner postpones a tax, he has, in effect, an interest-free loan from the government for the amount of the postponed tax.

Of course, if the tax is ignored rather than legally postponed, the studio owner will have not a loan but a debt to society that must be paid. That fine line—and the five to fifteen years saved—is why so many sound, recording and broadcast business owners rely on professional tax advisor. 

New Products

EFFECTS

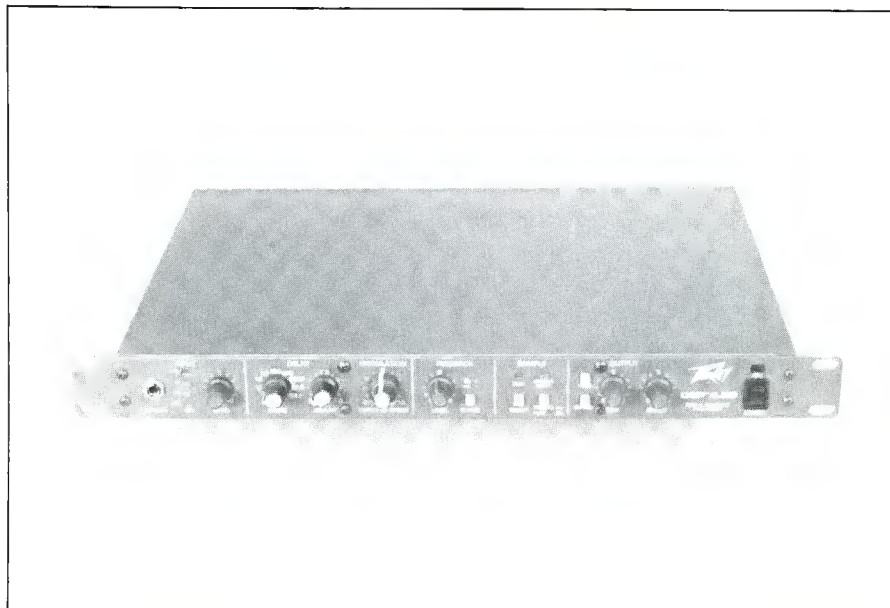
PROCESSOR/SAMPLER

● Peavey Electronics announced the release of the DEP 3.2S digital sampling processor, which uses proprietary VLSI (Very Large Scale Integrated) circuitry. This effects processor/sampler features 12-bit A/D/A conversion for low-noise operation, and provides a continuously adjustable delay range from 1.25 milliseconds to 3200 milliseconds. The audio bandwidth is 20 Hz to 10 kHz. The processor has stereo outputs, remote selectable hold and bypass functions, wide-ranging LFO modulation, dual-range input/output level selection, dual-range 12-bit sampling (up to 3.2 seconds), auto-trigger storage or loop playback modes and remote triggering capability.

Mfr.- Peavey Electronics Corporation

Price- \$349.50

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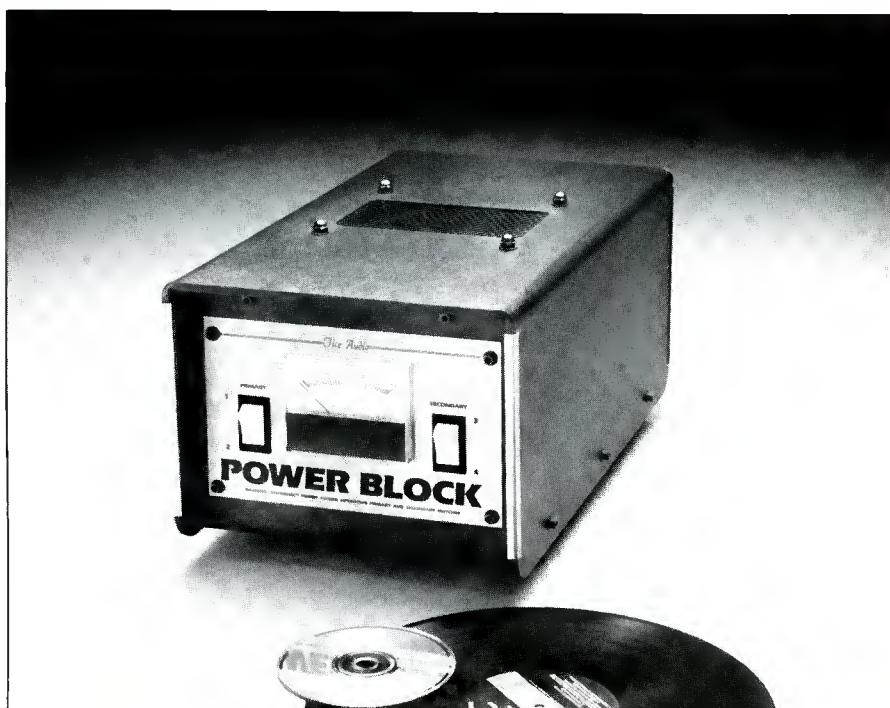
A.C. POWER CONTROLLER

● The Power Block is the first and only power line conditioner specifically designed for the needs of audio equipment. It has been engineered to remove the disturbances which affect audio equipment most. It is a complete a.c. power line purification and conditioning system. The benefits of the power block include stronger and more effortless bass, improved image capabilities and harmonic accuracy, less grit and grain, cleaner amplifier clipping and more effortless fatigue-free presentation of music. The unit's weight is 60 pounds.

Mfr.- Tice Audio Products Inc.

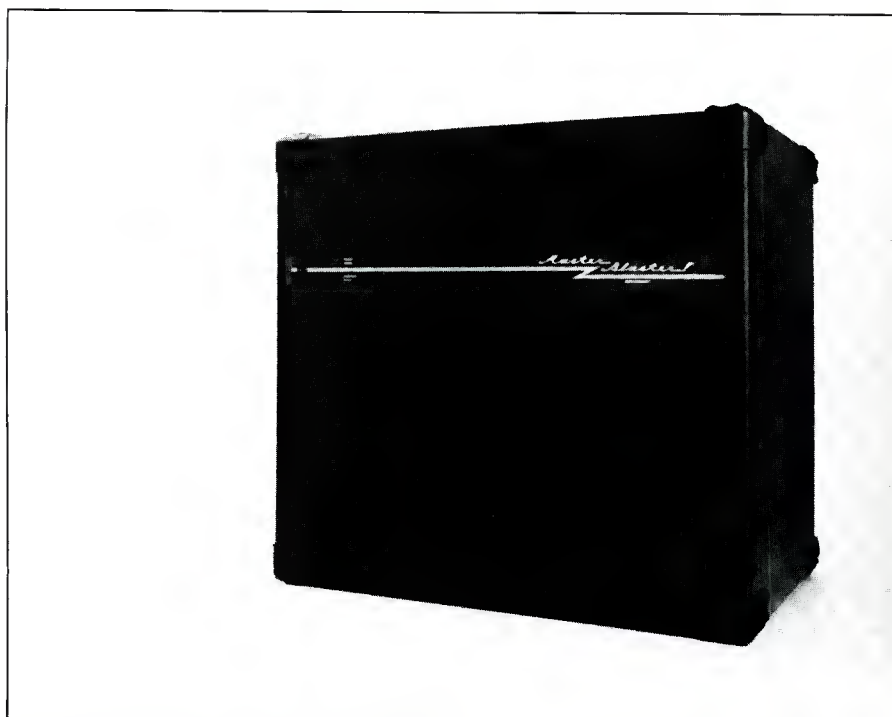
Price- \$1,150.00

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STUDIO MONITOR

● Applied Research & Technology introduces the Master Blaster, a pair of efficient 10-inch transducers with an 1-inch compression driver, each coupled to a constant coverage horn, the system seeks to eliminate the distortion associated with high sound pressure levels. Low end frequencies are handled by the port loaded enclosure that features a single, sturdy 18-inch driver. Both units that make up the system house their own crossover and amplifier internally mounted on a 3u high, 19-inch rack. A new generation of shock free amplifiers were designed capable of delivering extreme high peak power. The power amp module can deliver in excess of 1000 watts. The input signal is constantly monitored and compared with the acoustic output to guarantee faithful reproduction. Each cabinet weighs no more than 100 pounds and comes equipped with flying hardware and recessed handles. A delayed automatic level control built into each enclosure offers constant signal maintenance as opposed to limiting



or compression of the dynamic range.
Mfr.- Applied Research and Technology
Price- not available at press time

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EQ SYSTEM

● Altec Lansing Corporation announces the Programmable Equalizer System, comprised of five new products, two equalizers and three controllers, that working together constitute a complete programmable equalizer system.

The 8051A MicroAudio Autoprogrammer is a full-featured control unit. It provides a one-third octave equalizer, a controlled real time analyzer, a microprocessor, and a pink noise generator. This unit may be left in a system, but is usually used by the installer to set up one of the stand-alone equalizers permanently installed in the system.

A hand-held programmer, the 8055A MicroAudio Programmer allows the user to adjust each band of a stand-alone equalizer individually.

The 8061A MicroAudio PC Computer Control Adapter is a circuit board designed to be installed in an IBM PC, XT or AT type computer or a 100 percent compatible. The 8061A can program the equalizers as well as read the settings already stored in the equalizer memories. These settings can be printed out or saved on floppy disks.

A stand-alone, one-third octave

equalizer, the 8551A 28 Band/Single Memory MicroAudio Programmable Equalizer has a non-volatile memory so even if the a.c. power goes off, the settings programmed into the equalizer are "remembered." The unit is connected to one of the controllers only during the programming process.

The 8558A 28 Band/Multiple Memory MicroAudio Programmable Equalizer is identical to the 8551A with the exception that it can hold up to eight separate equalization curves that can be easily accessed from the front panel by an authorized user.

The programmable equalizer system is tamperproof, has multiple memory eq with read/write capabilities and is quiet with noise floors below -90 dBm. The system occupies a single rack space.

Mfr.- Altec Lansing Corporation

Price- \$900.00 (8551A)

\$3300.00 (8051A)

\$550.00 (8055A)

\$550.00 (8061A)

\$1100.00 (8558A)

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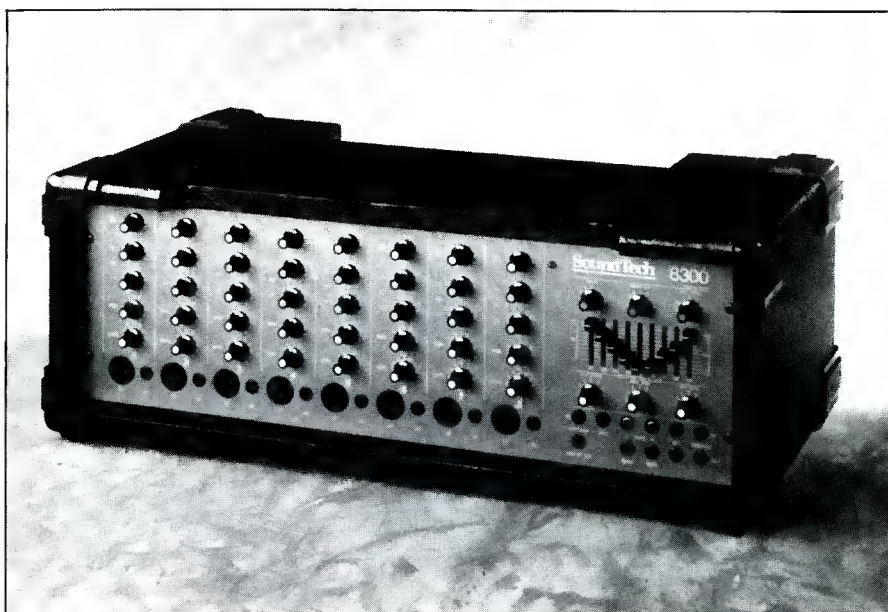
POWERED MIXER

● SoundTech has expanded its line of powered mixers with the addition of the 8300. The 8300 features eight channels and 300 watts of power. Each channel is equipped with balanced high and low impedance inputs, post effects sends, active high and low equalization, and separate pre-monitor sends. The 8300 includes a nine band graphic equalizer and extensive patching capabilities. Inputs are provided for a tape source and an external effect. The mixer, graphic eq and power sections can be used separately for greater flexibility.

Mfr. - SoundTech

Price - \$799.00

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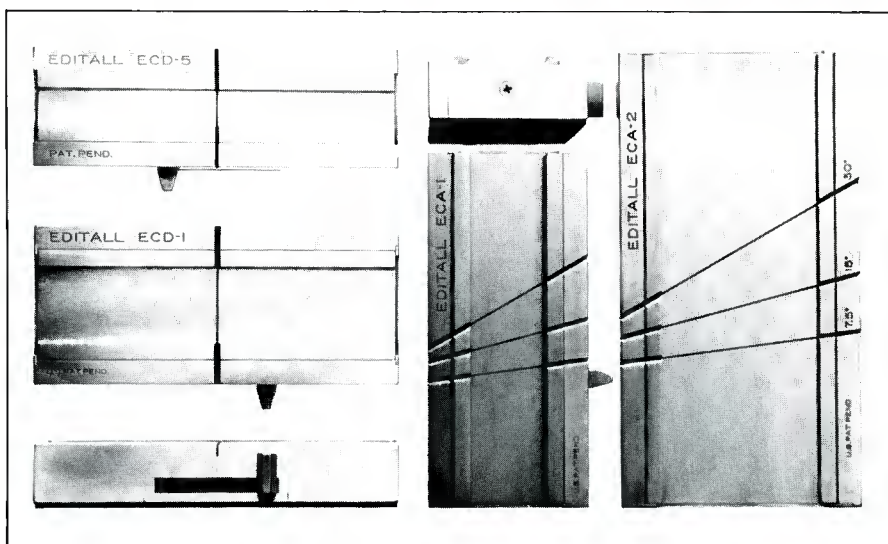


SPLICING BLOCK

● Editall has developed the "EC" series of magnetic tape splicing blocks. These precision blocks are for handling and splicing very thin, fragile tape as utilized in the various, new digital formats. Consisting of a flat bed, "Edge Clamping" configuration, these blocks eliminate the characteristic lifting and shifting of thin tape due to static attraction. This patented design also provides continuous, clear access to the tape for splicing, while it is held securely in the block. This tool permits the handling and blade splicing of these more demanding materials, resulting in clean, flat, curve free splices. The series includes the ECD-5 1/2-inch digital, ECD-1 1-inch digital, ECA-1 1-inch analog and the ECA-2 2-inch analog.

Mfr. - Xedit Corporation

Price - \$275.00 (ECD-5)



\$300.00 (ECD-1)

\$325.00 (ECA-1)

\$350.00 (ECA-2)

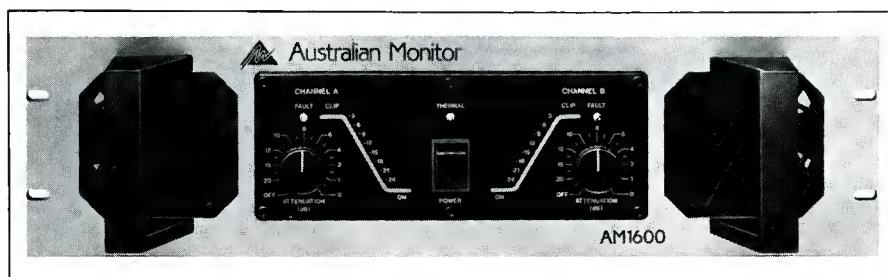
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POWER AMPLIFIER

● Graftons Sound introduces the AM 1600; an ultra-high power Mosfet amplifier that delivers 1600 watts at 4 ohms, 2200 watts at 2 ohms, and has the ability to operate at 2 ohms in high ambient temperatures. The unit has ultra quiet "on demand" cooling fans that when combined with custom designed alloy extruded heatsinks give efficient thermal operation. It has full protection features, slow turn-on and durable alloy chassis.

Mfr. - Graftons Sound Pty Ltd

Price - \$2950.00



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NAMM Round-Up

This year's Winter NAMM show has proven to be one of the largest in the history of NAMM. Mentioning all the products that are of interest to our readers would require an issue devoted to just those products. As a matter of fact, one could probably fill a book per month with nothing but new technology and the items that arise from the continuing course of advancement. However, we have selected a cross-section of new products introduced at the '88 Winter NAMM show which we feel will help to gain some insight into the present state-of-the-art and what the near future holds in store.

Welcome to our 1988 Winter NAMM Round-Up.

POWER AMPLIFIER

• Electro-Voice introduces the 7300 stereo power amplifier. It delivers 300 watts into 4 ohms and 200 watts into 8 ohms over the full audio bandwidth. In the mono bridge mode--activated by a mode switch on the back of the unit--the amp delivers 600 watts into 8 ohms. The optional APX plug-in module provides on-board biamp capability. The APX has Linkwitz-Riley, 24 dB-per-octave filter and 24 switch-selectable frequencies on one-third octave ISO centers from 50 Hz to 10 kHz. Features include LED clip indicator on each channel, LED protect indicator on each channel, detented pots on front panel, balanced XLR connectors, ring-tip-sleeve phone jacks, and protective brackets on binding posts.

Mfr.- Electro-Voice

Price- \$758.00

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CASSETTE RECORDER

Tascam has introduced America's first 8-track multi-track cassette recorder using standard audio cassettes. The rack-mount Tascam 238 Syncaset supplies twice the amount of tracks previously available in a standard cassette multi-track recorder, along with state-of-the-art control. Features include 3-3/4 in./sec. tape speed, full-function remote control, auto punch in/out, auto rehearse, dbx II noise reduction and MIDI (FSK) compatibility. Additionally, the ergonomically designed recorder is SMPTE-friendly, giving it the ability to lock up with other decks and synchronize with video. Another feature is a serial connector for external computer control and open architecture for future software develop-



ment. The soon-to-be-released Tascam MIDiiZER synchronizer will be an important part of the 238 system, allowing easy integration with MIDI instruments and SMPTE ma-

chine synchronization.

Mfr.- Teac Corporation of America

Price- \$2295.00

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WIRELESS MIC TRANSMITTER

● Shure Brothers Inc. has announced the introduction of the W15HT wireless microphone transmitter, a hand-held unit designed for use with the wireless microphone receivers. The mic transmitter is available in two versions: the W15HT/58, which is equipped with a Shure SM58 dynamic microphone element, and the W15HT/87, supplied with a Shure SM87 condenser element. Both the SM58 and SM87 heads may be used interchangeably with any W15HT transmitter. The W15HT's interior construction is surrounded by a ribbed ARMO-DUR housing that eliminates slippage. Heavy-duty construction gives the unit ruggedness and durability. Both of the microphone elements available for use with the W15HT also incorporate effective shock mounting for low-noise operation. The transmitter section operates at a single, crystal-controlled frequency in the VHF band between 166 and 216 MHz. A total of 15 frequencies, computer selected for interference-free operation, are available, and other frequencies can be ordered. This enables users to operate a number of wireless microphone systems in a single sound installation, simultaneously and without intermodulation problems. Operating controls on each unit include microphone on/off and power on/off with battery (9-volt) condition LED. Each mic transmitter is supplied with a swivel adapter for mounting on floor or desk stand, a transparent lockplate for covering the controls, and a small screwdriver for adjusting audio gain.



Mfr.- Shure Brothers Inc.
Price- \$750.00 (W15HT/58)
\$900.00 (W15HT/87)

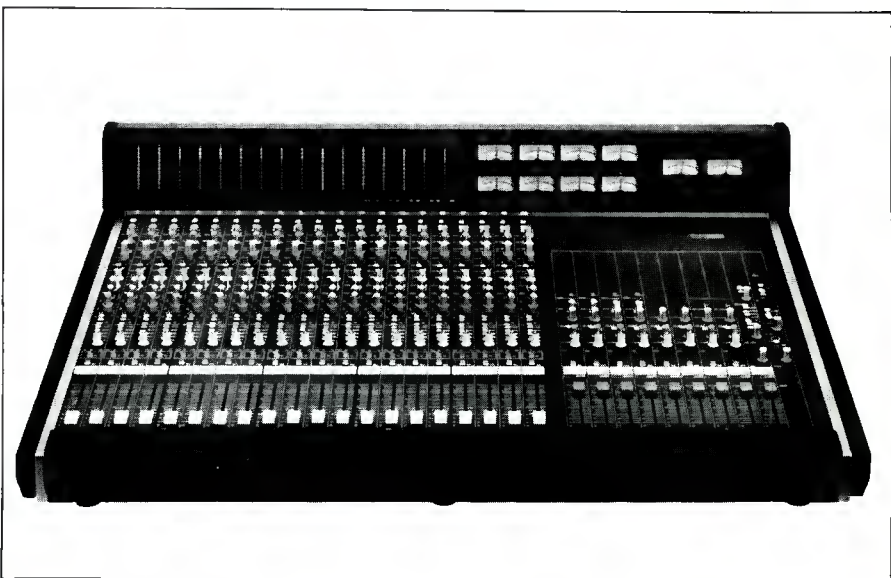
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MIXING CONSOLE

● Ramsa/Panasonic has announced the release of the WR-T820B recording console. It is an 8 bus console with flexible routing and switching that allows up to 48 inputs and 8 addressable auxiliary sends. High speed operational amplifiers are used at critical gain stages throughout the console's circuitry. New "MRP" 300,000 operation faders deliver smooth and accurate operation. Full function LED and VU metering enables metering of all inputs and outputs.

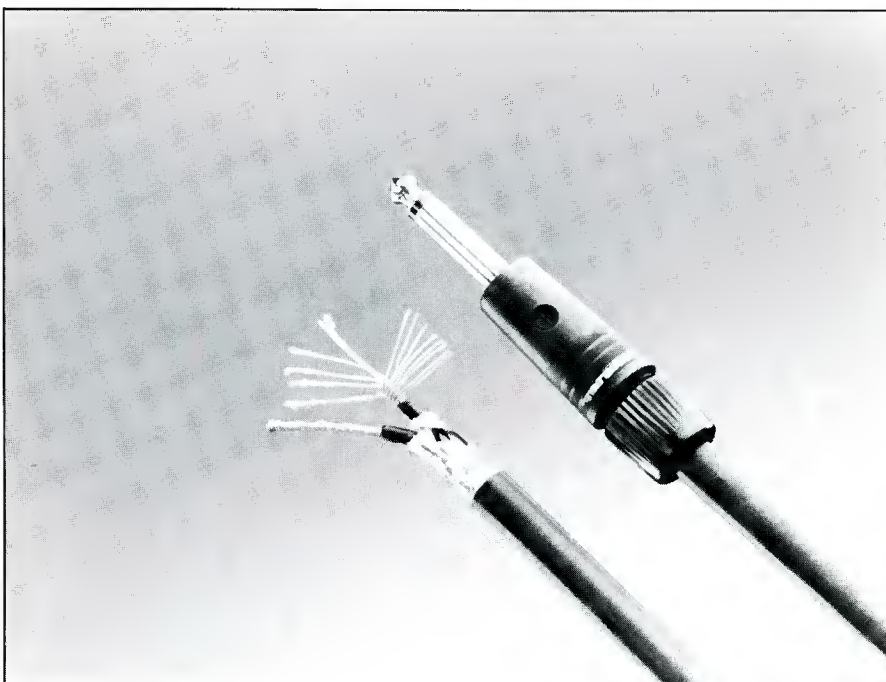
Mfr.- Ramsa/Panasonic
Price- \$8500.00

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INSTRUMENT CABLE

● Monster Cable introduces Pro-link Series 1 instrument cable. It incorporates "Bandwidth Balanced" construction. Three separate multi-gauge "wire networks" employ a single large conductor for bass, four medium-sized wires for the midrange, and 350 fine copper strands for the highs. Each network is then wound and "tuned" to control time delays. The cable employs "Micro-Fiber" dielectric insulation, wrapped around each conductor, to lessen the audible compression of high-level signals. It has a combination of heavy-duty foil and a 95 percent high-density braid to help reject annoying radio frequencies and electromagnetic interference, and low-noise CP conductive plastic layers to eliminate handling noise. The cable also comes with a "Duraflex" jacket that is UV-stabilized and resistant against abrasion, chemicals and temperature extremes. Precision black chrome-plated 1/4-inch phone connectors with "Collet-type" strain relief pro-



vide reliable connections.

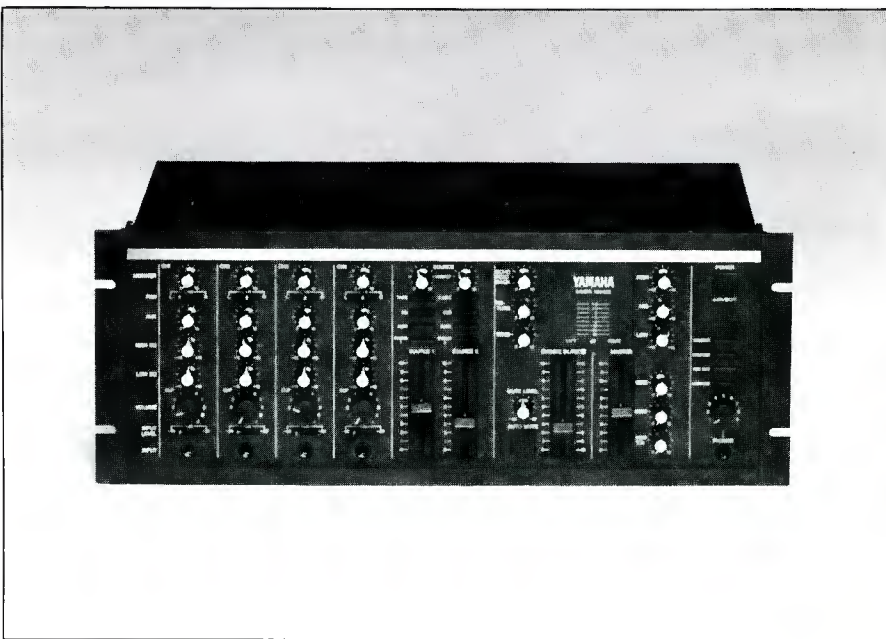
Mfr.- Monster Cable

Price- \$6.00 to 12.00 per foot

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MIXER

● Yamaha Music Corporation USA announces the MV422 multi-source mixer, specifically designed for performance and versatility in churches, discos, clubs, auditoriums and similar institutions. In a single rack-mount unit, the mixer offers four microphone/instrument input channels, each with selectable input sensitivity, eq, auxiliary send control, selectable panning and monitor level control. Two additional Source channels permit input and mixing of audio and A/V sources such as tape, CD, phono, video disc, VCR and others. The Source 2 Video Disc and VCR input selectors switch the video as well as the audio. A single Source Balance fader facilitates smooth, single-handed cross-fades between the two Source input groups. There is a BBD echo system built-in, with an Echo Feedback control for setting any degree of echo to match the source. An Auto Mute system provides fully automatic gain riding. A comprehensive monitor system provides versatile monitoring of all system signals.



Mfr.- Yamaha Music Corporation,
USA

Price- \$795.00

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Classified

FOR SALE

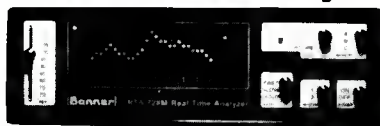
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People, Places... & Happenings

● **Richard A. Ferguson** has been named chairman of the **National Association of Broadcasters' Radio '88 Steering Committee**. He was appointed by NAB President Edward O. Fritts and Joint Board Chairman Wallace Jorgenson. The Management, Programming, Sales and Engineering Convention will be held September 14-17 at the convention center, Washington, D.C. The first committee meeting will be held in early February to review Radio '87 activities and to consider plans for Radio '88.

● The **Audio Engineering Society Educational Foundation** has announced the opening of its 1988 educational grant program for university studies with emphasis on audio topics. Awards, for graduate students only, are made annually, and successful applicants may request a one-time renewal of their grants. Additional information and applications are available from the AES Educational Foundation, 60 East 42nd St., New York, NY 10165.

● Plans are now underway to move **Saki Magnetics'** manufacturing operation to new headquarters. The decision to move was made in order to expand the operation into a new custom designed facility with more production area. The move should be completed by the first week in April. The new address is: 26,600 West Agoura Road, Calabasas, CA 91302, (818) 880-4054.

● **Ken Kitamura**, chairman of **Mitsubishi Electric America (MEA)**, the parent company of **Mitsubishi Pro Audio Group (MPAG)**, has announced the promotion of **Tore Nordahl**, the current president of MPAG, to Director of New Business Development at MEA. **Mr. S. Miyata** will be the new president of MPAG. He will be responsible for coordinating the world wide operation of MPAG, and controlling the expansion of the company.

● Audio pioneer **Bob Liftin** passed away on January 8, after a long and distinguished career in professional audio. Suffering from cancer, he continued working until the end, engineering for the CBS New Year's show broadcast from New York's Waldorf Astoria. Mr. Liftin began his career as a teenager with CBS, engineering audio for radio soap operas and then made the transition to audio for live television. He launched Regent Sound Studios in the late '50s, where top records originated over the years. He was a founding member of SPARS, and served as president and chairman of the board.

"Bob was always five years ahead of the industry," recalled Sandi Morrof, general manager of Regent and A2 (TV audio assistant) for the past seven years. "He was instrumental in developing SMPTE lock up and synchronization of audio and video and also paved the way for much of the computerized, digital storage that is used today. He was a tremendous human being—the man was a champ."

As an audio consultant and engineer, Mr. Liftin was associated with some of the most demanding and prestigious broadcast events: The 1986 Liberty Weekend, The Tony Awards, Radio City Music Hall live broadcasts, Live Aid from Philadelphia, The Jerry Lewis Telethon and served as audio consultant for Saturday Night Live since its inception.

"He was in love with audio," said Dave Teig, studio manager at Servisound, NE coordinator of SPARS, and longtime friend. "He was an innovator and tireless in his pursuits."

Bob Liftin is survived by his wife Susan, sister Marilyn, son Jim and daughter Elise.

● **Delos International, Inc.**, the Southern California-based record company, announced that it will relocate its corporate headquarters to Chatsworth, California. An independent producer of classical and jazz recordings, Delos is timing the move to its fifteenth anniversary celebration, according to **Amelia Haygood**, Delos' president. The move will permit expansion of all phases of operation. The facility will also include new digital audio editing suites, equipped with the Sony PCM-1630 digital mastering system. The new space houses corporate offices, production suites and a distribution center. The new address is: 9244 Jordan Ave., Chatsworth, CA 91311.

● **Stanley N. Baron**, managing director in the Technical Division of NBC, Inc., has been elected Engineering Vice President of the **Society of Motion Picture and Television Engineers (SMPTE)**. Beginning in January 1988, Baron will assume responsibility for the supervision and coordination of the work of the nine technical committees of the Society. Baron has been involved in the design and development of digital television systems for more than 21 years. For the SMPTE, he has had a key role in the development of standards for digital video, and is currently co-chairman of the SMPTE Standards Committee and Society Engineering Director for Television.

● The **L.A. Record Plant** has sold 50 percent of its operation to UK-based records, music and entertainment company, **Chrysalis Group plc**. On December 8, 1987, the multi-million dollar studio complex officially became "Record Plant, a Chrysalis Group plc Company." **Chris Stone**, who will remain president of the new division, stated that Record Plant had been looking for the best way to expand their operation, and Chrysalis had been looking for a facility base on the West Coast. They expect growth to extend beyond audio into video and film post-production.

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NARADA MICHAEL WALDEN

Narada Michael Walden is a world-class drummer, keyboardist, singer, composer and performer. If there's a musical role he can't handle, no one has thought of it yet. As a producer, he's turned out such hits as Aretha Franklin's "Freeway of Love" and Whitney Houston's "How Will I Know" (which he wrote and co-wrote respectively). As a drummer, he's played jazz, fusion, and rock with the likes of John McLaughlin, Jeff Beck, and Weather Report, and R & B with Rick James and Teena Marie.

Narada is an extraordinary musical craftsman. He demands the very best from his music and his equipment. His choice in personal multi-track recording gear is AMR. Naturally.

NARADA MICHAEL WALDEN recent awards:

1986 ASCAP Songwriter of the Year:

"Freeway of Love" Aretha Franklin

1986 ASCAP Song of the Year:

"How Will I Know" Whitney Houston

1986 Billboard's Producer of the Year

1985 Grammy



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